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By Luke McCready

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By Mike Rothacher

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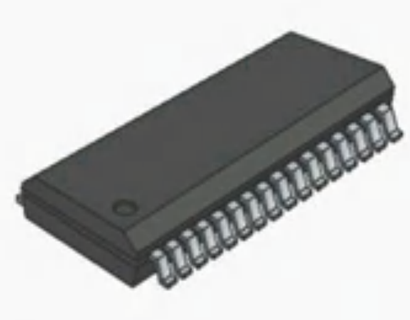
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The FuzzMeasure layout is clean: The left side shows recorded measurements to choose from, and the right contains the selected graphs, displaying the data of selected measurements. The top of the interface shows how much gain the connected microphone is receiving at any given time (see **Photo 1**).

There are only four buttons on the interface. The first is to take a new measurement; the second is to add a new graph to view measurements; the third includes the minimal amount of I/O settings, including sample rate and the number of channels being recorded. And, the last button opens up the Measurement and Graph sub-panels with additional options and information.

Graphs look crisp and high-resolution, and colors are automatically assigned. Exporting graphs as an image is a great feature: The visual is re-rendered according to the dimensions you choose, and there's even an option to internally render at 2× the resolution and downscale the final result. I rendered a couple of graphs at 3840 × 2160, and they looked great.

Measurements are taken by sine wave sweeps, a far more accurate method over real-time analysis for most purposes. FuzzMeasure includes the ability to export its measurements as a .csv file containing magnitude for 16,383 different frequencies, regardless of the sample rate. Technically, this means that a measurement of the audible range is more detailed when created at 44.1 kHz than 96 kHz.

Rather than share endlessly about its feature set, well-described in its help documentation, allow me to take you on a tour of how I've found FuzzMeasure useful in my time with it. And the timing is just right, as I'm still dialing in my new studio space.

A Simple Frequency Test

When I built my power amplifier with the Audio Society of Minnesota, I opted to install two sets of inputs, one with a capacitor for filtering out low frequencies and one without, though I've never measured at what frequency the filtering begins and how aggressive it is. The perfect job for FuzzMeasure? I rearranged two super-chunk bass traps and an absorption panel to make a "poor man's anechoic chamber" with a speaker firing into the absorption toward the measurement mic (see **Photo 2**), and in just a couple minutes of testing, I had results (see **Figure 1**).

The relevant numbers are that the capacitor rolls off the sound by 1 dB at 150 Hz, 3 dB at 80 Hz, and 6 dB at 49 Hz. That could be relevant for some people when dialing in their sub's low-pass filter or subtly shaping the tone of a monitor's low-end.

Unique Approach to Comb-Filtering

Ever in the pursuit of purer sound, I searched for weeks for a 2′ × 4′ desk with a roll-out keyboard tray so I could cover the surface with an absorption panel to combat comb-filtering, the ugly acoustic phenomenon of high frequencies doubling and negating each other as the result of minutely delayed reflections. I measured with and without my absorptive surface to see if it was making a difference (see **Figure 2**).

The results were not what I was expecting: The frequency response was different, but not a net positive or negative. However, the clarity improved enough to make the DIY project worthwhile (see **Figure 3**).

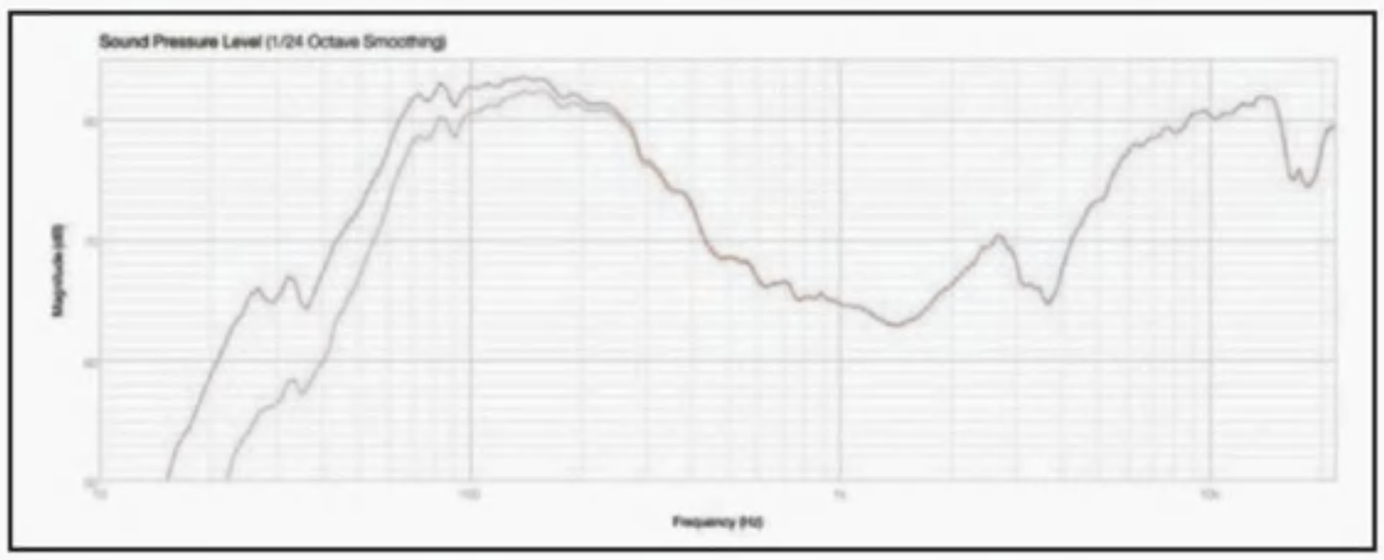


Figure 1: The red trace is without low-frequency filtering, and the orange trace is with.

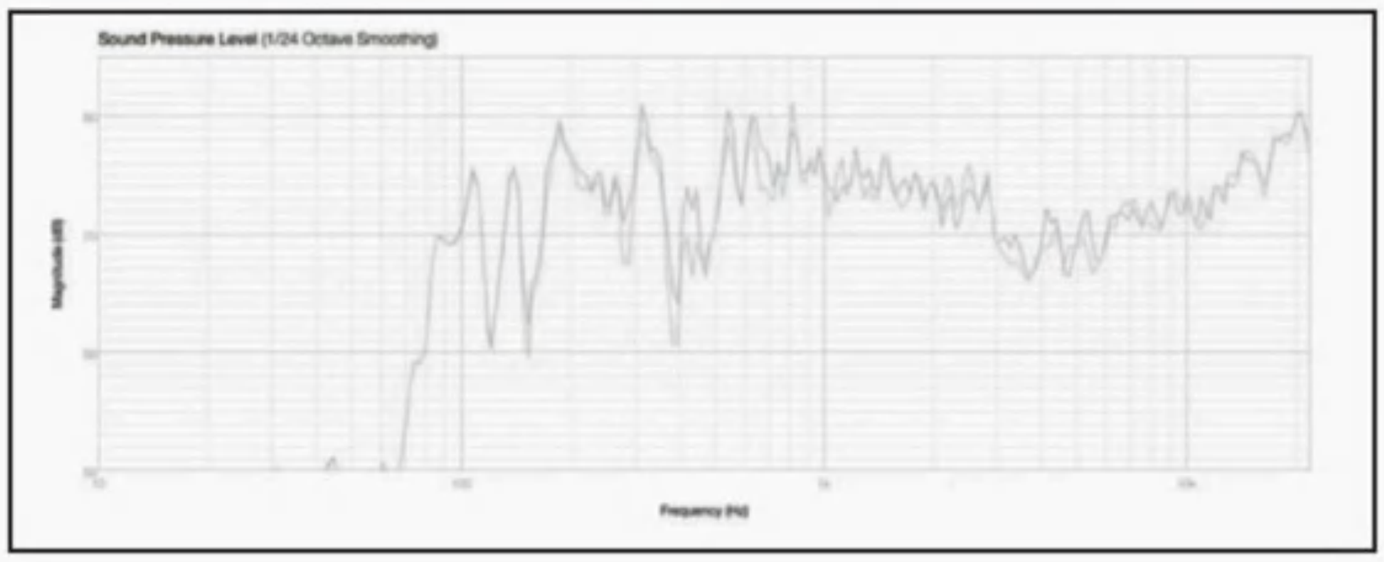


Figure 2: The green trace shows my desk-covering absorption panel at work, and the blue trace shows without. I had expected more benefit for eliminating the primary source of comb-filtering in my room.

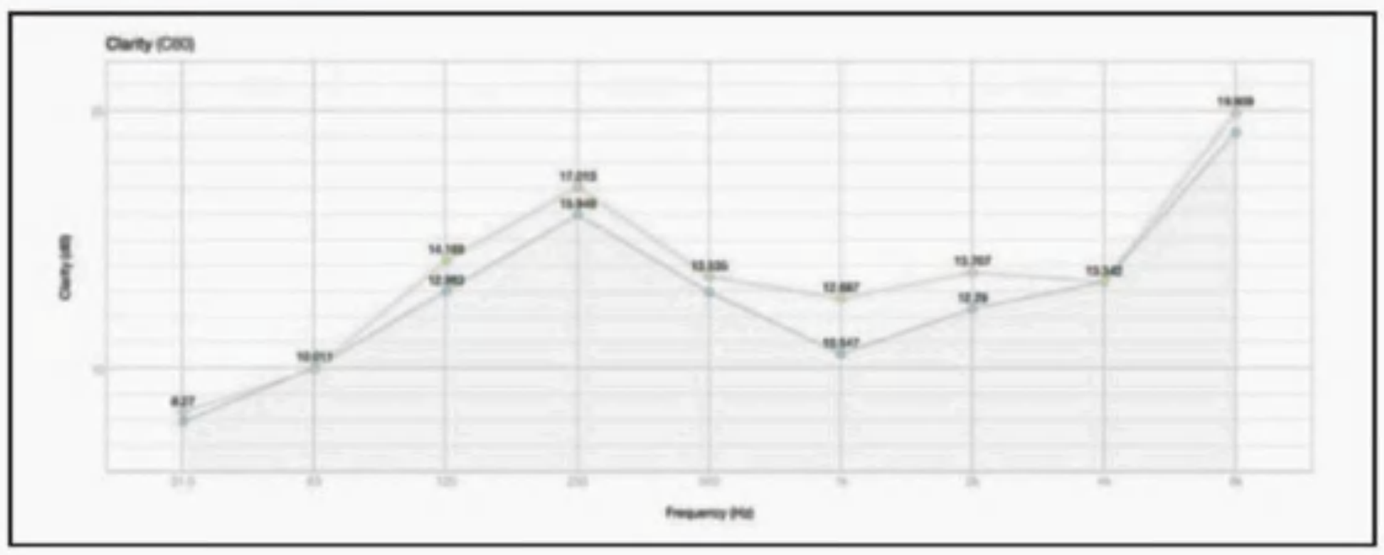


Figure 3: The gains in clarity by overcoming comb-filtering are more obviously beneficial than the changes in frequency response.



Photo 3: Measuring my DIY speakers (a), against an Elac Uni-Fi UB5 (b), and Frazier La Fiesta (c).

A Small Speaker Shootout

Next, I put my DIY monitors to the test against an Elac concentric three-way and a pair of vintage Frazier one-way speakers (see **Photo 3**). When measuring audio, it's important not to apply too much smoothing, and to keep the graph's scale relevant (see **Figure 4**).

After viewing a more reasonably detailed graph, none of the speakers look perfect, and the frequency differences begin to stand out (see **Figure 5**). This graph also includes first and second harmonic distortion, and each of the three speakers has its trouble areas with distortion. Something worth noting: The frequency curves of the speakers are heavily correlated from 100 Hz to 300 Hz—this speaks to acoustical problems in my room.

Measuring for Crossover Settings

Finding the right frequency to cross over signal from your monitors to your subwoofer is important: Set it too high and your system will sound uneven and bloaty; set it too low and deep bass may sound defined, yet there will be a sizable

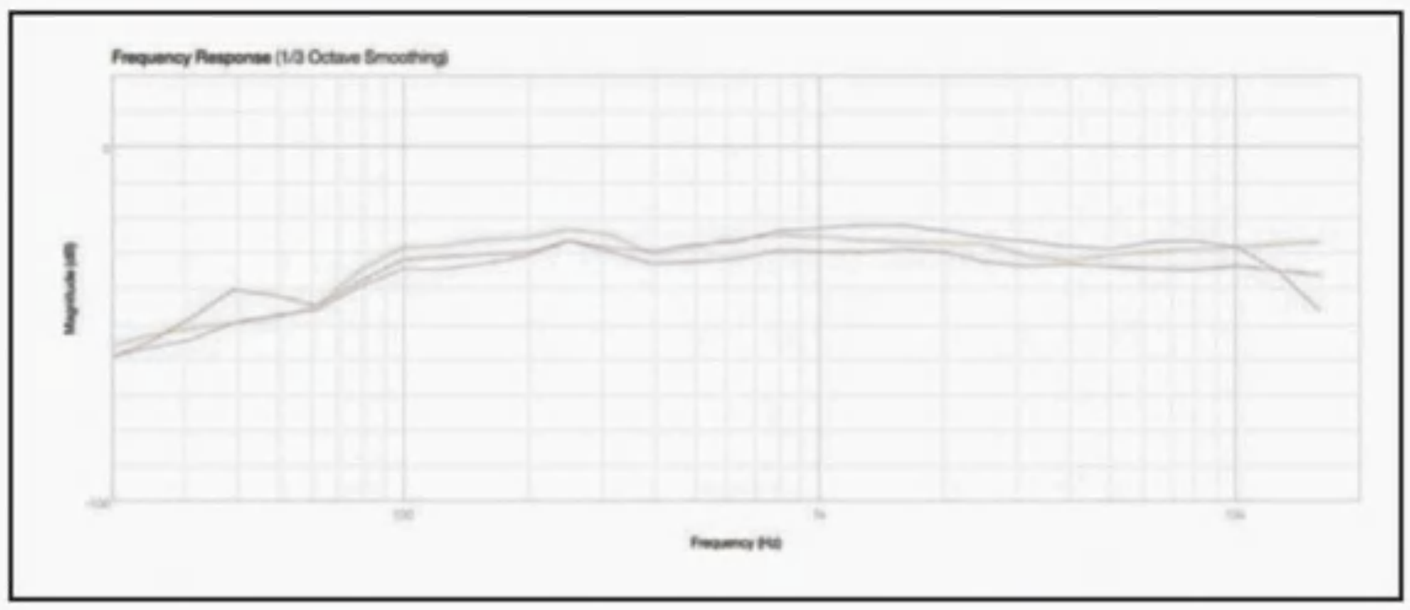


Figure 4: At a glance, these speakers all look to have a fantastic frequency response. But with 1/3 octave smoothing and the graph scaled to show 120 dB of possible data, the visuals are quite misleading.

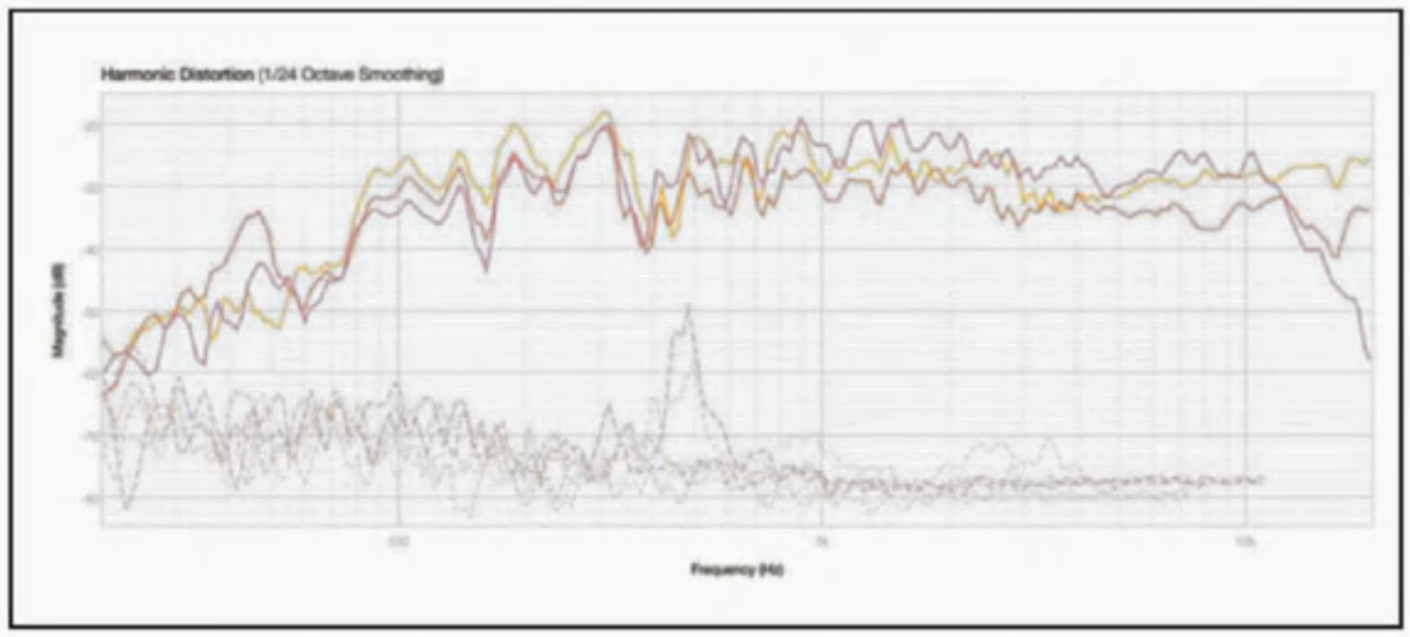


Figure 5: The Fraiser, in purple, has very poor high-frequency response and substantial distortion over the frequencies the human ear is most sensitive to. The Elac, in red, has quite a bit of distortion in the midrange and a surprising frequency bump at 55 Hz.

About the Author

Luke McCready is an audio engineer, a technical writer, and a music lover based out of Denver, CO. He graduated with a degree in audio engineering from North Central University with a passion for audio and a fascination for speaker setup and acoustic treatment.

gap in the mid- and upper-bass frequencies. Can FuzzMeasure be used as a tool for finding the best crossover frequency? Let's find out (see **Figure 6**).

I had expected a dramatic upper-bass bloat or a deep mid-bass hole, but the results were not as cut-and-dry. It's easy to forget that crossovers are gradual and continue to interact much further along the spectrum. Following the purple trace, the 40 Hz crossover contributed the least to the 100 Hz bump but dropped far too low in the 70 Hz dip. And following the sea-green and navy blue traces, we see that 140 Hz and

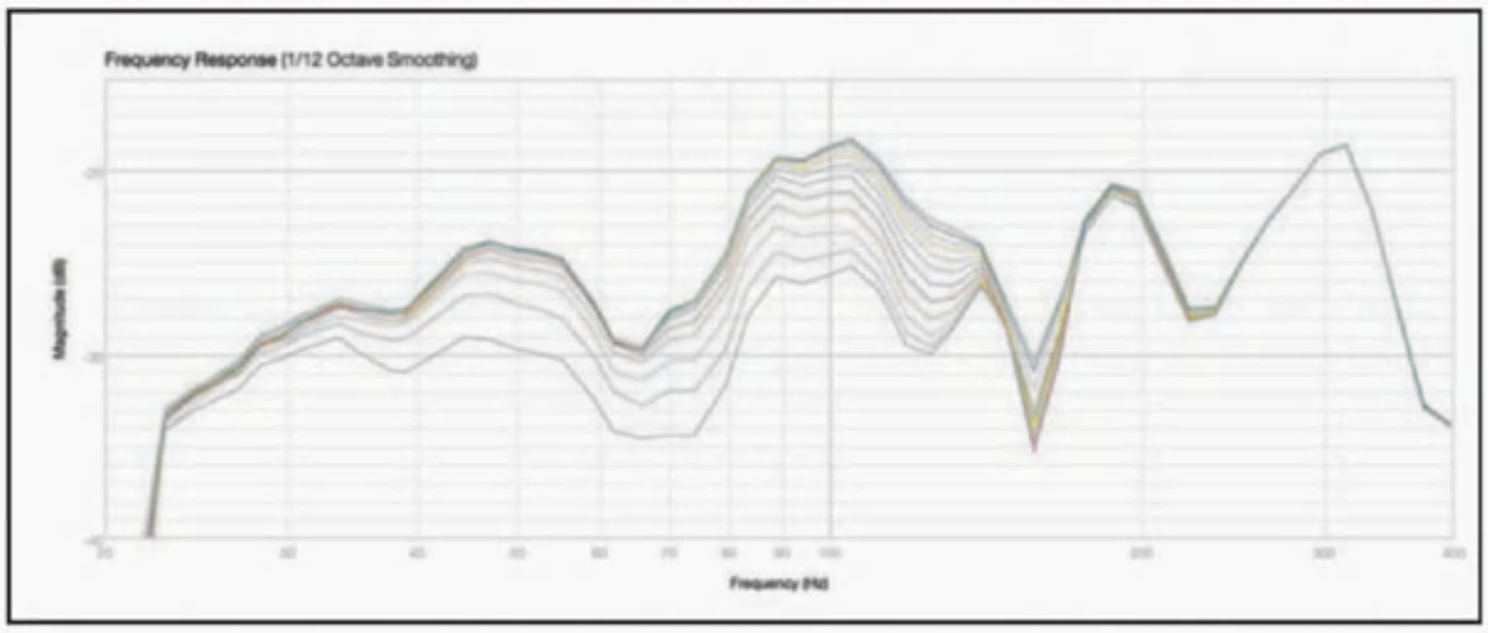


Figure 6: I tested four subwoofers with two monitors; crossover frequencies ranging from 40 Hz to 150 Hz.

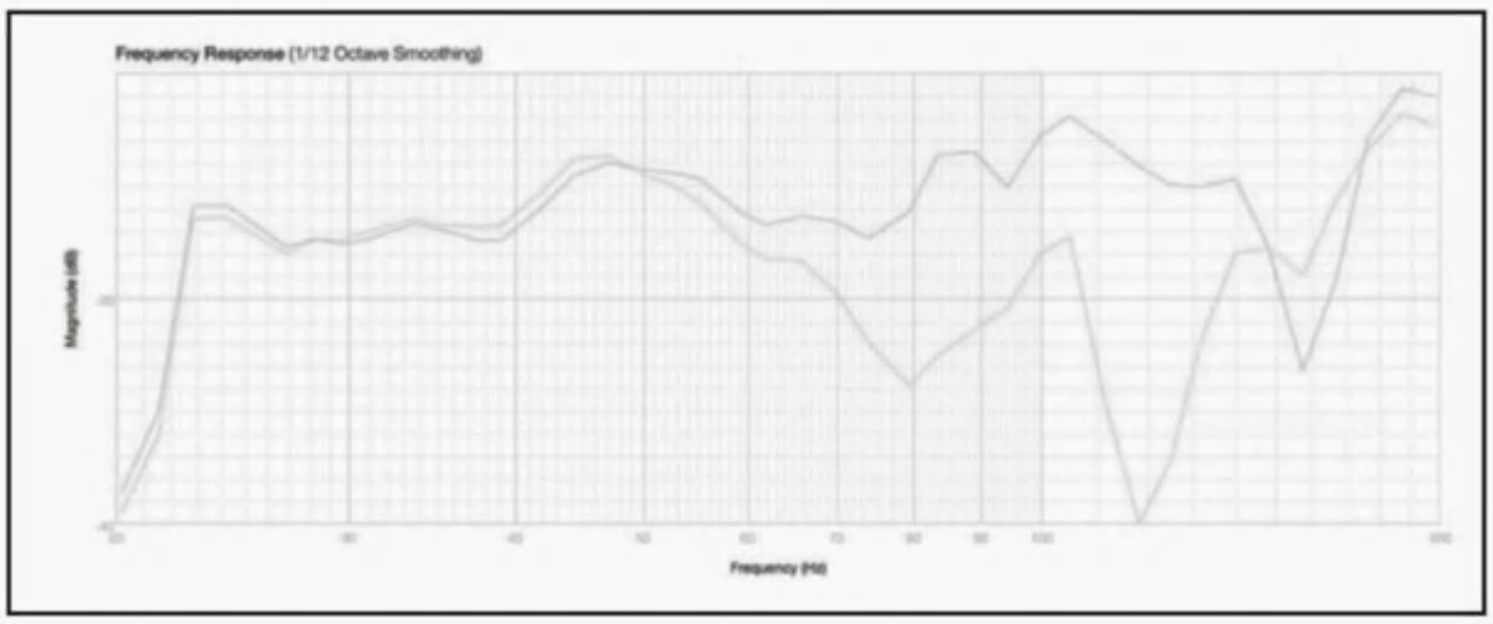


Figure 7: Can FuzzMeasure help you spot out-of-phase speakers? Definitely. The green trace shows the massive loss of mid- and upper-bass when the subs are out of phase with the monitors.

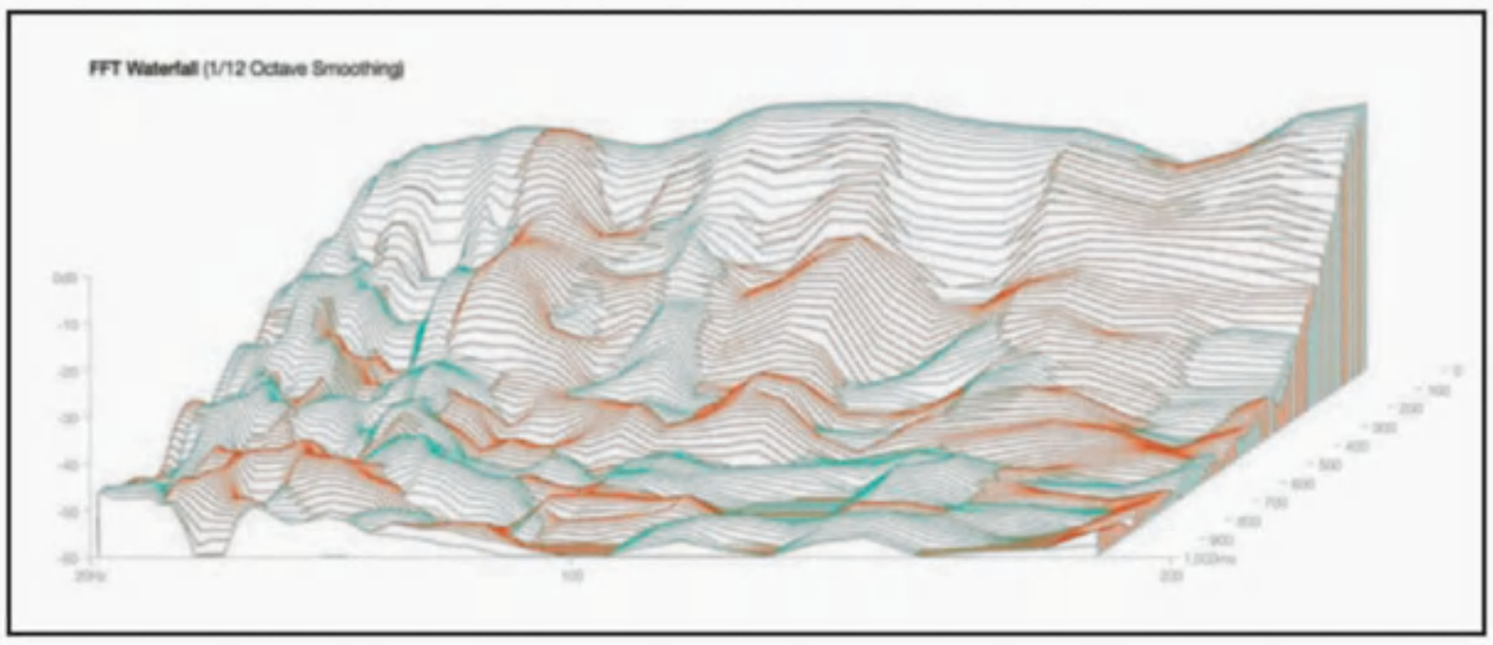
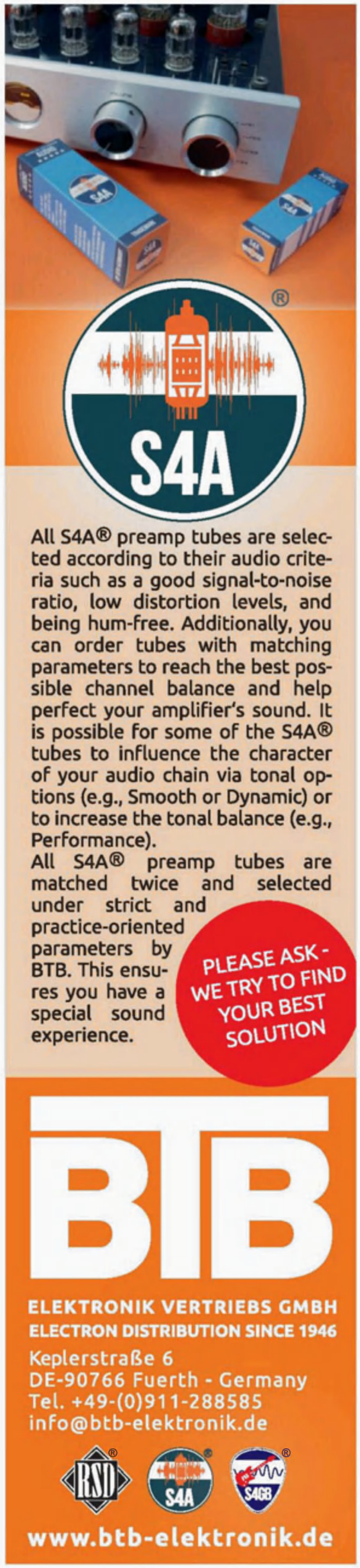


Figure 8: Bass response without absorption in red and with absorption in teal—proof that absorption can even out frequency response, quicken sound decay for most frequencies, and even make certain frequencies louder at the listening position by absorbing frequency-canceling reflections.



150 Hz have the opposite problem. All of these options would benefit from corrective equalization (EQ), and assuming EQ, most of the crossover frequencies in the middle of the graph would suffice. However, my setup with sealed monitors and dipole subwoofers is unusual—most monitors and subs are ported and experience a much steeper 24 dB/octave roll-off, making the right crossover frequency both more important and easier to find.

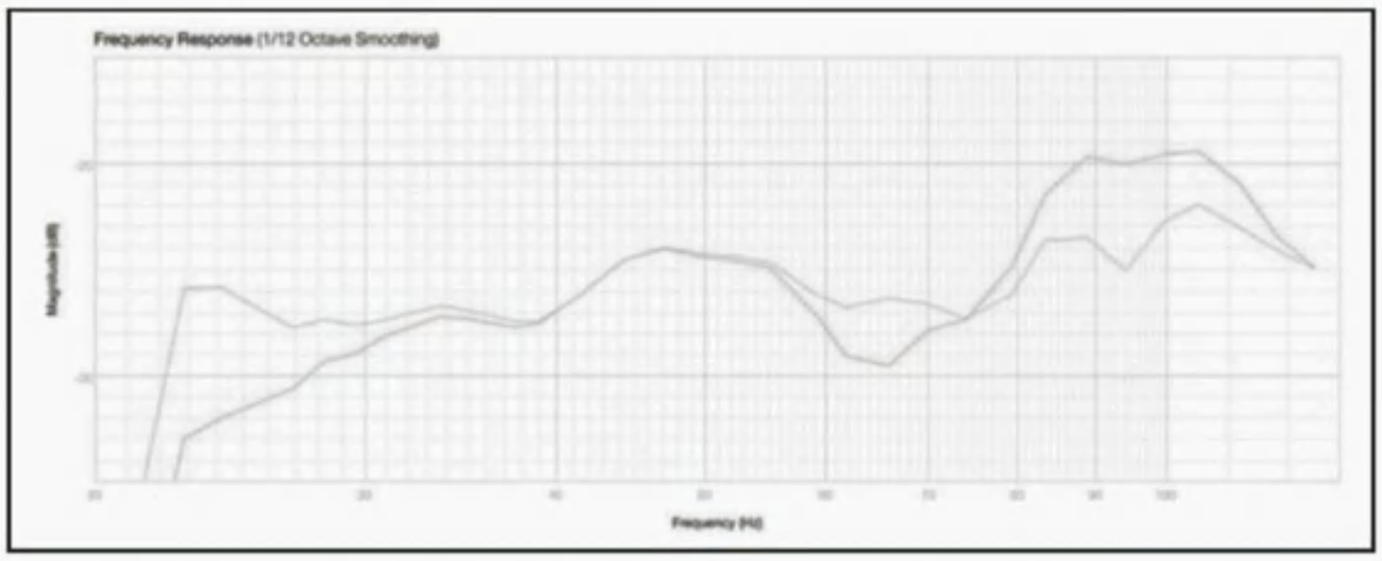


Figure 9: The orange trace shows the frequency response of the subs in my room. By comparing in FuzzMeasure, I was able to improve my subwoofer frequency response with a little manual EQ—see the blue trace.

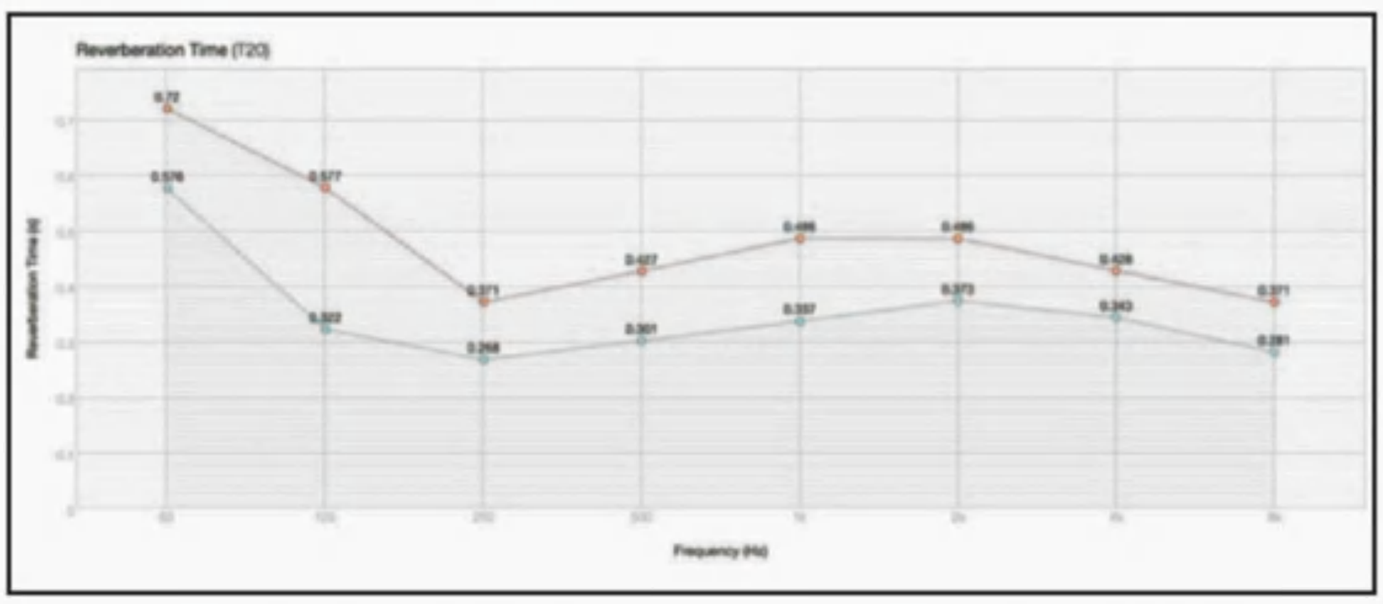
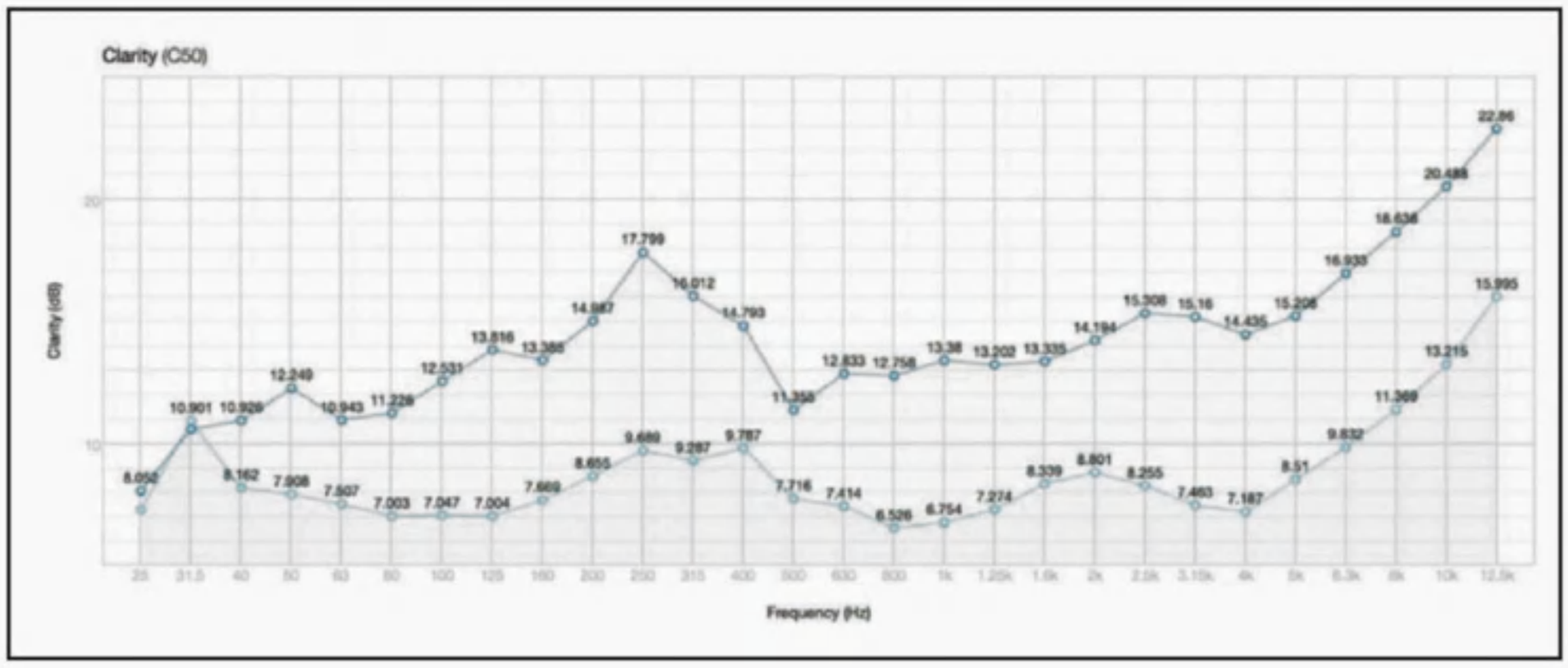


Figure 10: The red trace represents frequency-dependent reverberation time without acoustic treatment, and the teal line shows the tightened reverberation time with acoustic treatment.

Figure 11: The light blue trace is without acoustic treatment, and the dark blue trace is with. I'd much rather listen to a system with that clarity than without.



FuzzMeasure has many additional uses for subwoofer setup, including checking phase (see **Figure 7**).

Measuring Bass and Bass Traps

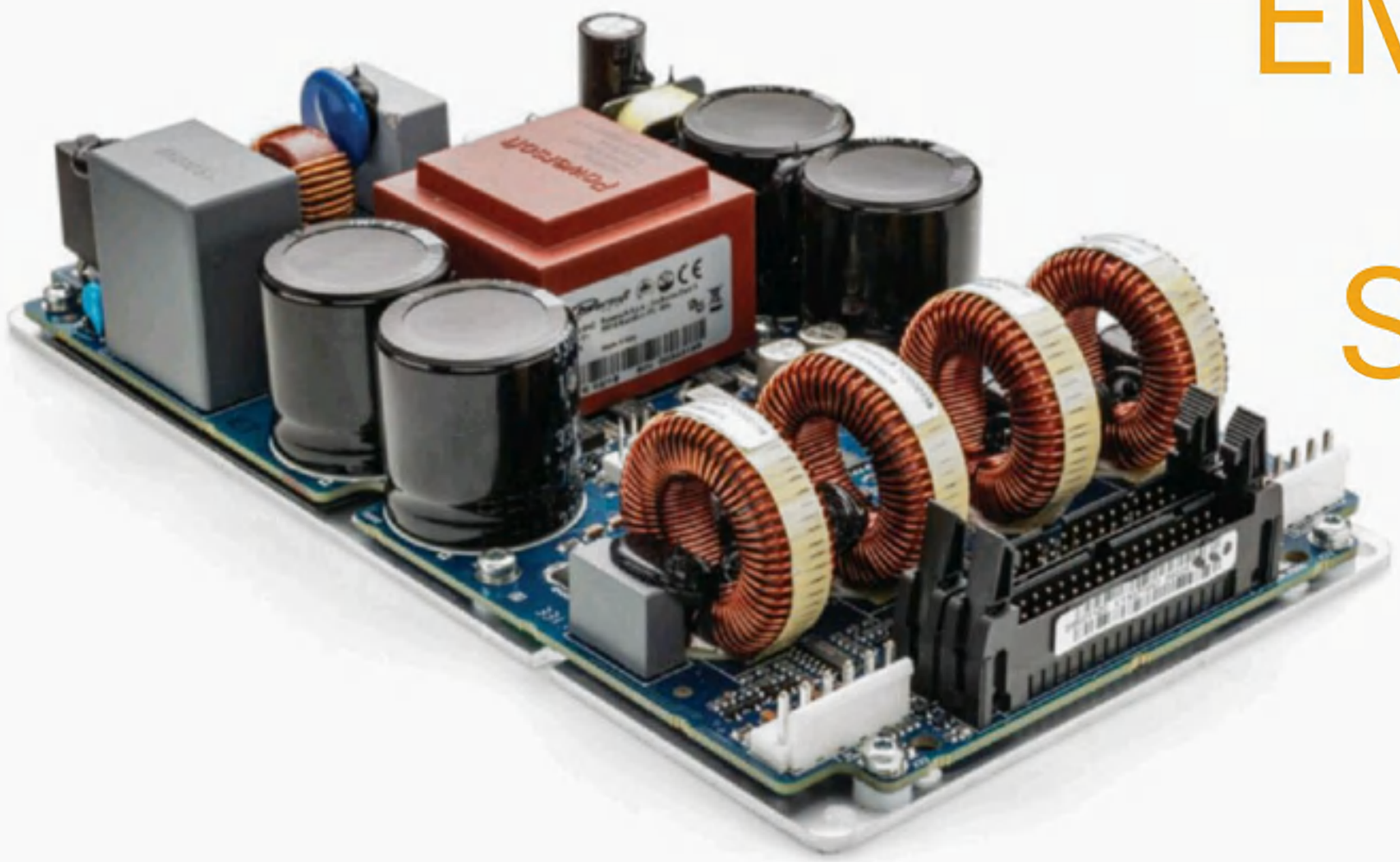
Virtually all rooms benefit from acoustic treatment. For those of us outside of large venues, that means copious absorption, and as thick as possible to be useful at low frequencies. I built my absorption using Owens Corning 703 that comes in 48" × 24" sheets. I made six full-size broadband absorption panels at 6" thick and three more at 4" thick. Also, I cut additional full-size sheets into four triangles each, measuring 24" × 24" with a 33" hypotenuse. I stacked these triangles into wooden frames 54" tall and placed one of these "super-chunk bass traps" into each of my room's four corners. Does it make a difference? Let's start by looking at my bass response with and without just the four corner traps (see **Figure 8**).

I used a waterfall graph for this test, to demonstrate sound decay over time. The red ridges sloping down demonstrate a longer decay without absorption. The teal upper ridge shows a more even bass response and even increased bass loudness at the listening position. Curiously, absorption seems to have lengthened the decay time of some low frequencies, though at much less relevant levels than the decay times reduced.

But the bass response still isn't pretty. Can FuzzMeasure inform users as to where and how to use EQ to correct frequency response? It can't work in tandem with certain EQ units as REW does, nor can it provide correction itself like Sonarworks Reference or Dirac Live. But I did use FuzzMeasure's graphing to decide which areas need EQ. Boosting EQ to work against frequency nulls is a dangerous game and only to be attempted if you have the amplifier headroom. I applied EQ with my

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Application Examples



MiniDSP 2×4, and using just a few bands in less than 10 minutes, I was able to substantially cut down on a peak, partially fill in a null, and extend the deep bass output of my subs (see **Figure 9**) by measuring, making an EQ change, measuring again, and so forth.

Broadband Absorption Measured

What happens when we add my broadband absorbers into the room? I placed two at each of the side-wall first reflection points and another covering my desk. I placed the rest of the panels around the walls of my 20′ × 12′ room.

The frequency response improved, but less than I would have guessed. Yet, frequency response is not the full story: Reverberation time is reduced at all frequencies (see **Figure 10**), giving a tighter, clearer quality to audio. And clarity is substantially improved at all but the very lowest frequencies as well (see **Figure 11**).

Grading Third-Party Equalization

My biggest recent studio upgrade was adding Sonarworks Reference 4. Instead of providing you bountiful data, Reference measures the frequency response of your system and corrects for it (see **Photo 4**). Measurements are obtained with the provided calibrated measurement mic and a clever software tool, and correction is applied with a desktop application that affects the audio of all programs and/or a DAW plug-in to use on your master bus.

Reference provides the equalization, but what does that correction look like under FuzzMeasure's microscopic analysis? Used without any acoustic treatment, it helps, but not nearly as much as I would have expected. Peaks are reduced, nulls are somewhat filled, high frequencies are more balanced, and bass response is slightly extended (see **Figure 12**). That's great, but Reference only



Photo 4: Sonarworks Reference 4 Systemwide module is capable of equalizing for custom room measurements and tailoring the audio to taste.

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Focusrite, PLC | https://focusrite.com

Loss Senderos Studios, LLC | https://lossenderosstudio.com

Micromega MyDac | https://micromega.com

MiniDSP 2x4: used for subwoofer crossover and dipole correction; not in the signal chain for the monitors
miniDSP | https://www.minidsp.com

Monitors: sealed DIY speakers in curved cabinets, MISCO 6.5″ woofer and SB Acoustics 1″ tweeter

FuzzMeasure 4.2.2
RØDE Microphones | https://www.rode.com
RØDE Test | https://www.rodetest.com

Sonarworks Reference 4, version 4.4.3.2 and Sonarworks Reference calibrated measurement microphone
Sonarworks | https://www.sonarworks.com

Subwoofers: 4× cabinets, DIY dipole design by Siegfried Linkwitz, 2× 12″ drivers per cabinet

SuperMegaUltraGroovy | https://supermegaultragroovy.com

TC Electronic BMC-2 volume control

reduced a passable ±8 dB frequency response to a mediocre ±6 dB response.

If your theory is that Reference isn't able to fill nulls that would better be served by absorption, repeating the test with all acoustic treatment back in the room tells a different story (see **Figure 13**). I don't understand why the response without EQ drifted to ±12 dB, and EQ can only rein it in to ±6.5 dB. However, all the clarity benefits of adding treatment are there, and room decay is reduced as expected (see **Figure 14**).

A little more testing revealed the mystery: Reference 4 decides its EQ curve based on the summing of 37 sine sweeps in unique locations around the listening position. I couldn't remember all 37 locations, but I remembered 25 of them, measured with FuzzMeasure, and averaged the results (see **Figure 15**). There's the ±2.5 dB I would expect from a sufficiently advanced EQ. And without the room-induced dramatic frequency swing between 100 Hz and 110 Hz, the response is more like ±1.75 dB—truly exceptional.

A few caveats, though: I wish Reference could correct for a smaller listening area, which would further flatten the true frequency response in the

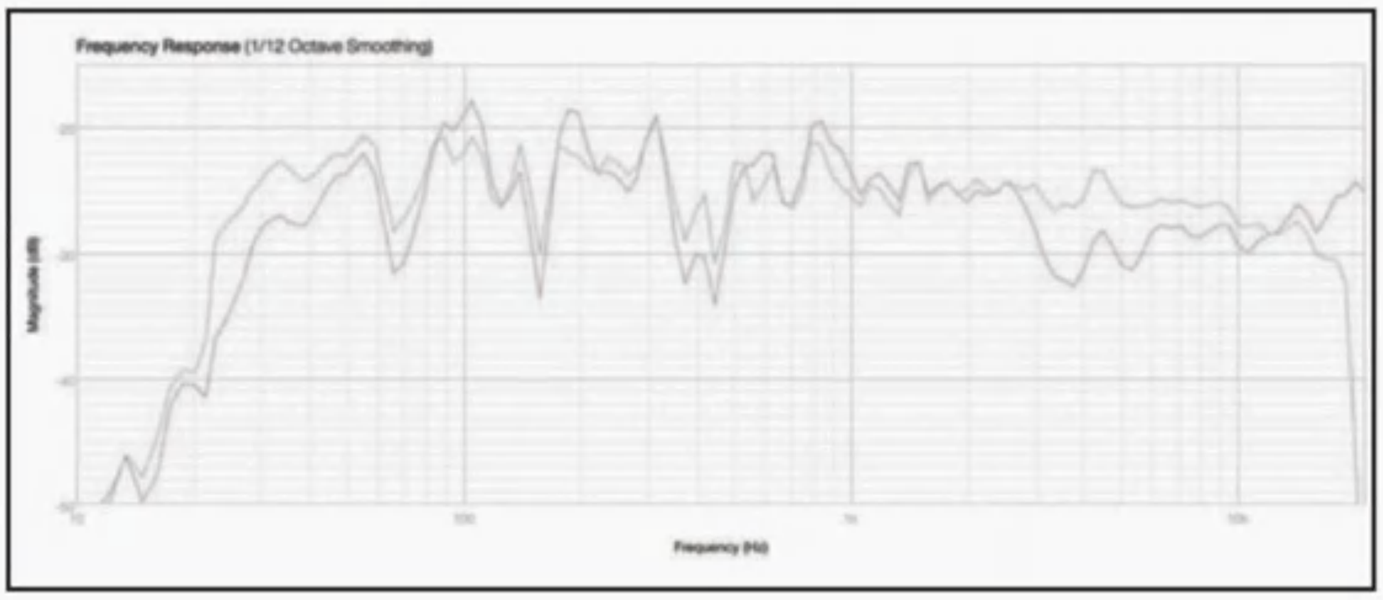


Figure 12: The red trace shows a full-range system without EQ and the blue trace is with EQ in an acoustically untreated room.

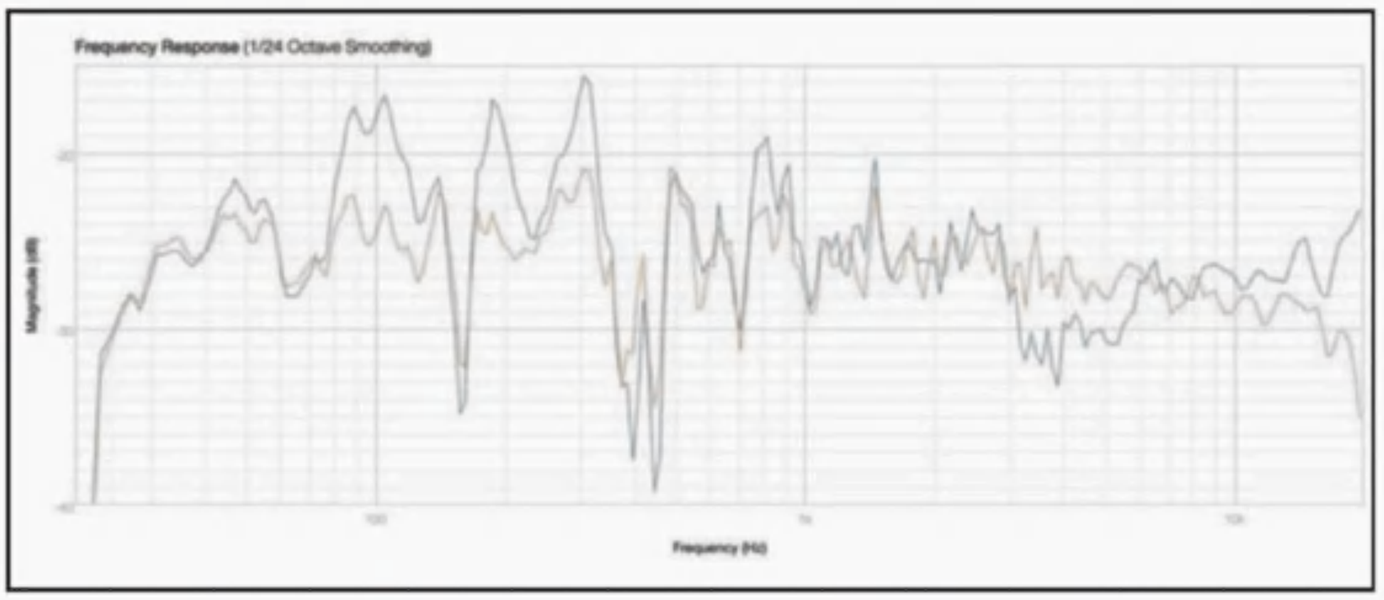


Figure 13: A tighter crop of the data range makes the frequency response look exaggerated on first glance. The blue trace is without EQ and the orange trace is with EQ, in a room with a good amount of broadband absorption.

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mixing seat. And I'm forming the opinion that the acoustic treatment I built for my previous studio location is not adequate in this new, larger room. I would not have thought of these without FuzzMeasure.

Slightly Buggy

I struggled to get my mic preamp gain loud enough for FuzzMeasure without clipping. Eventually, I discovered that I couldn't keep my mic plugged into my interface's second input without offering FuzzMeasure another mic in the first input. It took some hair-pulling to get FuzzMeasure to work at 96 kHz, wrangling sample rate settings in three places and performing a couple of computer restarts—I can't be sure if this is the fault of FuzzMeasure or my interface, though I've never had such troubles with the interface before in Windows.

Try as I might, I can't get useful data out of the Difference tool. In theory, I should be able to use

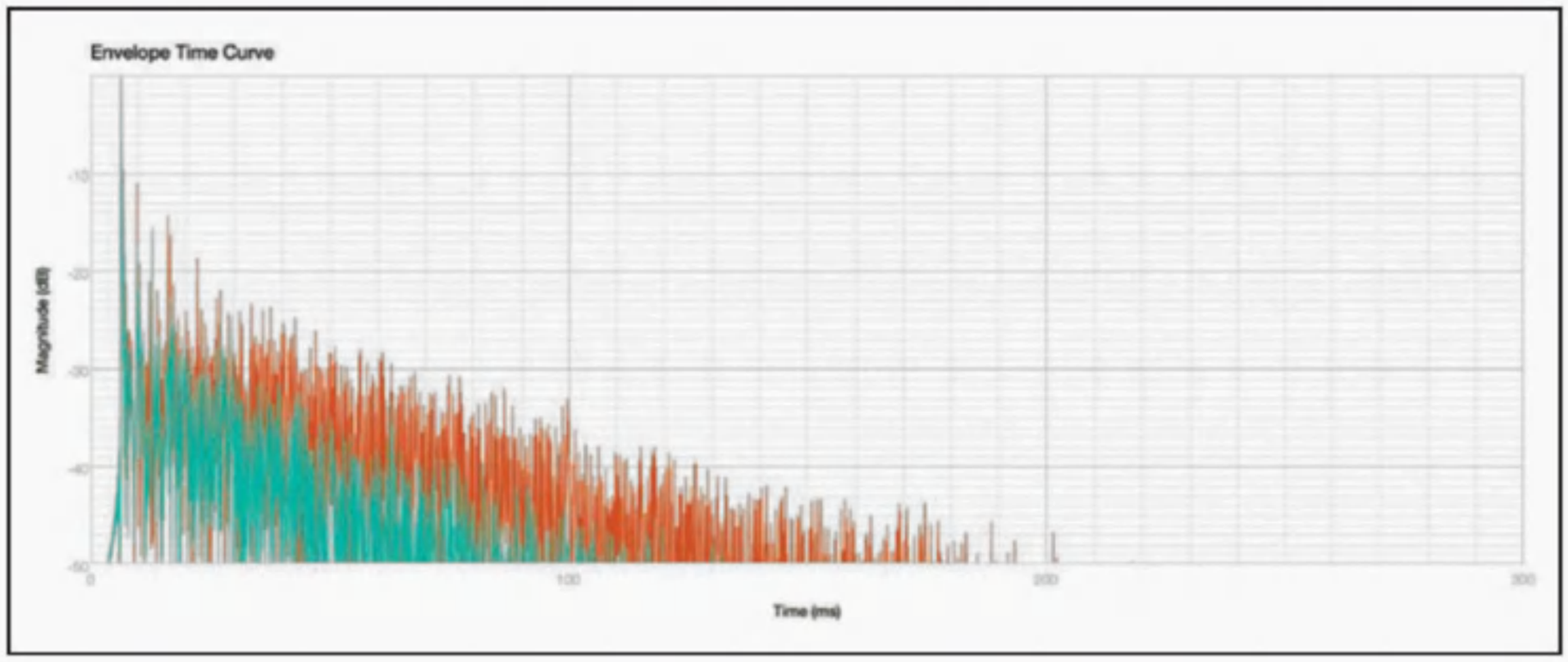


Figure 14: The Envelope Time Curve graph shows the decay of all audio as a whole. The red trace is without acoustic treatment, and the aqua trace is with treatment.

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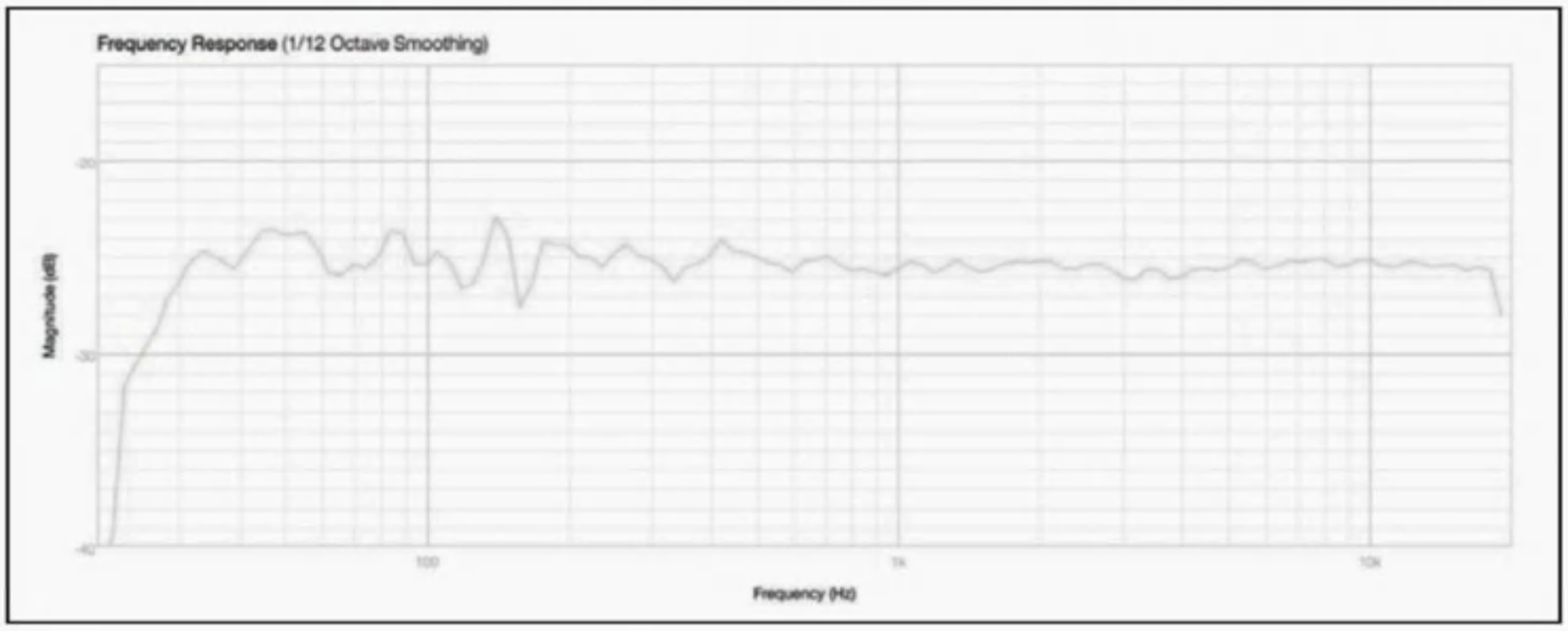


Figure 15: Averaging many measurements with FuzzMeasure more closely resembles Reference's anticipated output after its own many measurements.

it to see how the frequency of the left and right channels differ by 3 dB to 10 dB. Instead, I get readings that separate the channels by more than a million decibels.

Conclusion

FuzzMeasure is a powerful program that's also simple to use. I found it incredibly helpful in answering questions that have long plagued me, it showed the strengths and weaknesses of my current acoustic treatment in a way I hadn't seen before, and it even revealed a significant shortcoming in my favorite room correction software.

The clean-looking, high-fidelity graphs are perfect for anyone publishing their work, professional or hobbyist. And FuzzMeasure has additional uses beyond the scope of this review, including direct equipment measurement.

I get the sense that FuzzMeasure has not grown a great deal since it was acquired two years ago, but I look forward to Røde measurement mics coming in the future, and I like the idea of Røde taking on the world of measurement equipment and software just as it did on-camera microphones.

As FuzzMeasure does not provide any function beyond measuring and displaying information, it's a powerful tool for select uses only. Considering that, and its macOS-only status, you'll know whether or not you have need of it. And if you do, I heartily recommend it. ax

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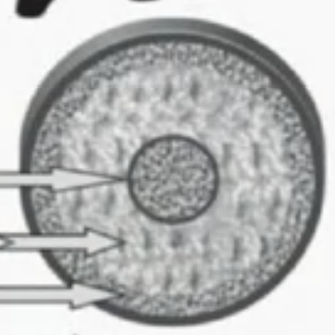


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Stop Worrying About Coupling Capacitors!

By
Ethan Winer
(United States)



Photo 1: I designed the tester's front panel in Microsoft Visio, then printed it onto card stock and glued it to the metal case.

Ethan Winer said he often sees arguments in hi-fi forums and Facebook audio groups about the importance of high-quality capacitors in the signal path. It's true that capacitor quality varies wildly among the various types, and the main problem with poor quality capacitors is they can add distortion to audio passing through them. But for inter-stage coupling capacitors, this may not be as big a problem as many people think. Winer built a simple device that measures capacitor distortion using a USB sound card and free software, and this article details the results of his tests with four different capacitor types.

Audio circuits require a DC voltage to operate, so all devices have either a battery or a power supply that converts the AC voltage from the wall outlet to a DC voltage needed by the circuits. If you power an audio circuit from AC, the 60 Hz (or 50 Hz) mains frequency will modulate the audio and appear at the output as hum.

Solid-state devices operate on low voltages, typically 15 V or less, while tube circuits require hundreds of volts. The internal input and output points of many circuits are not centered at 0 VDC, so coupling capacitors are used between stages. This is especially needed with tube devices, but also with many solid-state circuits that use discrete transistors instead of op-amps. So if the at-rest DC voltage at the output of one internal stage is 5 V, but only 2 V at the input to the next stage, a coupling capacitor is placed in the signal path to allow audio to pass without upsetting the DC voltage difference.

A capacitor is constructed from two metal plates in close proximity but not quite touching. The larger the plate surfaces, and the closer the plates are to each other, the greater the capacitance will be. One common construction uses parallel strips of metal foil, with thin plastic insulation between the strips. The assembly is rolled up to save space, then dunked into goo that hardens. Since the metal plates aren't electrically connected, DC can't get through. However, current can flow through a capacitor when the applied voltage changes. Although there's no physical connection through a capacitor, a light bulb attached to a battery through a capacitor will flash briefly when first connected.

Therefore, a capacitor passes AC because the voltage keeps changing. And high frequencies get through more easily than low frequencies because the voltage changes more quickly. You can think of a capacitor as a frequency-selective resistor, and, in fact, that's exactly what it is! Sending audio in series through a capacitor creates a high-pass filter because less signal passes through at low frequencies. On the other hand, wiring a capacitor as a short circuit across the audio signal creates a low-pass filter with higher frequencies reduced in level. The cut-off frequency of these filters depends on the capacitor's value—its capacitance—along with the destination load resistance.

One problem with capacitors is they can distort the audio passing through them. But this happens only when the signal level changes through the

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capacitor. At very high frequencies, all of the audio gets through; and at very low frequencies, none of the audio gets through. Distortion occurs only when some of the audio passes and some is blocked. So it's important for capacitors in the signal path to have low distortion, but this applies mostly for equalizers and other filters where the capacitor's loss occurs within the audible pass-band. Coupling capacitors in series between stages of an audio circuit generally have a large enough value to roll off starting below 20 Hz. Since little audio voltage is lost across a coupling capacitor at the higher audible frequencies, in theory their distortion should not be a factor. This is exactly what I set out to prove or disprove with my tests.

The main contributor to capacitor distortion is its voltage coefficient. This is a measure of how much the capacitance changes as the applied voltage varies. Ideally, a capacitor should have the same amount of capacitance no matter how much voltage is present across its terminals. Another contributor to capacitor distortion is its dielectric absorption, which is a "memory" effect. After discharging a capacitor fully, some DC voltage reappears soon after because part of the charge was stored within the capacitor's dielectric—the technical term for the insulating material between the metal plates.

Capacitor distortion greatly varies from very low with Mylar, polystyrene, and mica types to unacceptably high with some electrolytics, and even higher for ceramic types. Here, the capacitor "type" identifies its dielectric material. High-density ceramic capacitors that pack a lot of capacitance into a small physical size are even worse, and also suffer from the piezoelectric effect. This generates an audible noise when the capacitor is tapped or subjected to vibration.

Measuring Capacitor Distortion

As part of my work as an acoustic consultant, I've come to rely on the Room EQ Wizard (REW) software. This "donationware" program measures every aspect of room acoustics (e.g., frequency response, reverb decay times, and much more). It also includes a distortion analyzer to measure the distortion of loudspeakers. Coupled with a high-quality sound card (e.g., the Focusrite Scarlett 8i6), REW can be used to measure the distortion of passive components (e.g., capacitors and transformers) and even active devices such as equalizers and preamps. There are other audio measuring programs such as the popular RightMark Audio Analyzer, which is meant for exactly this sort of testing. But I already have REW and know it very well.

Measuring capacitor distortion is more complicated than simply sending audio through the component being tested. Capacitors being measured must "drive" a resistor load connected to ground to force a voltage drop across the capacitor, in order to emulate how they're used in audio circuits. And testing different value capacitors requires changing the load resistance to keep the cut-off frequency within the audible band. This way we can see how the distortion varies both below and above the cut-off frequency. A 22 µF capacitor driving a 1,000 Ω load has a low-frequency cut-off at 7 Hz, which is too low for REW to display. Further, testing polarized capacitors requires a DC bias voltage large enough to ensure the polarity is never reversed by audio present across the cap. The largest test signals I used were REW's swept sine waves at slightly less than 2.2 V_{PP}, so a 1.5 V AA battery was sufficient for my tester. That is, the reverse portion of the sine wave polarity was 1.1 V or less, leaving a 0.4 V safety margin.

Photo 1 shows the front panel of my device, which is simply card stock glued to an aluminum project box with 3M spray photo mount. I used a Zero Insertion Force (ZIF) socket for the capacitors being tested because it's a lot easier to insert and remove components than with a normal IC socket. Plus, I already had it. This socket accommodates both thick and thin wires, which is needed with the range of capacitor values I tested.

The capacitor being tested and the Load Impedance potentiometer plus a 620 Ω "limit" resistor form a high-pass filter, with the

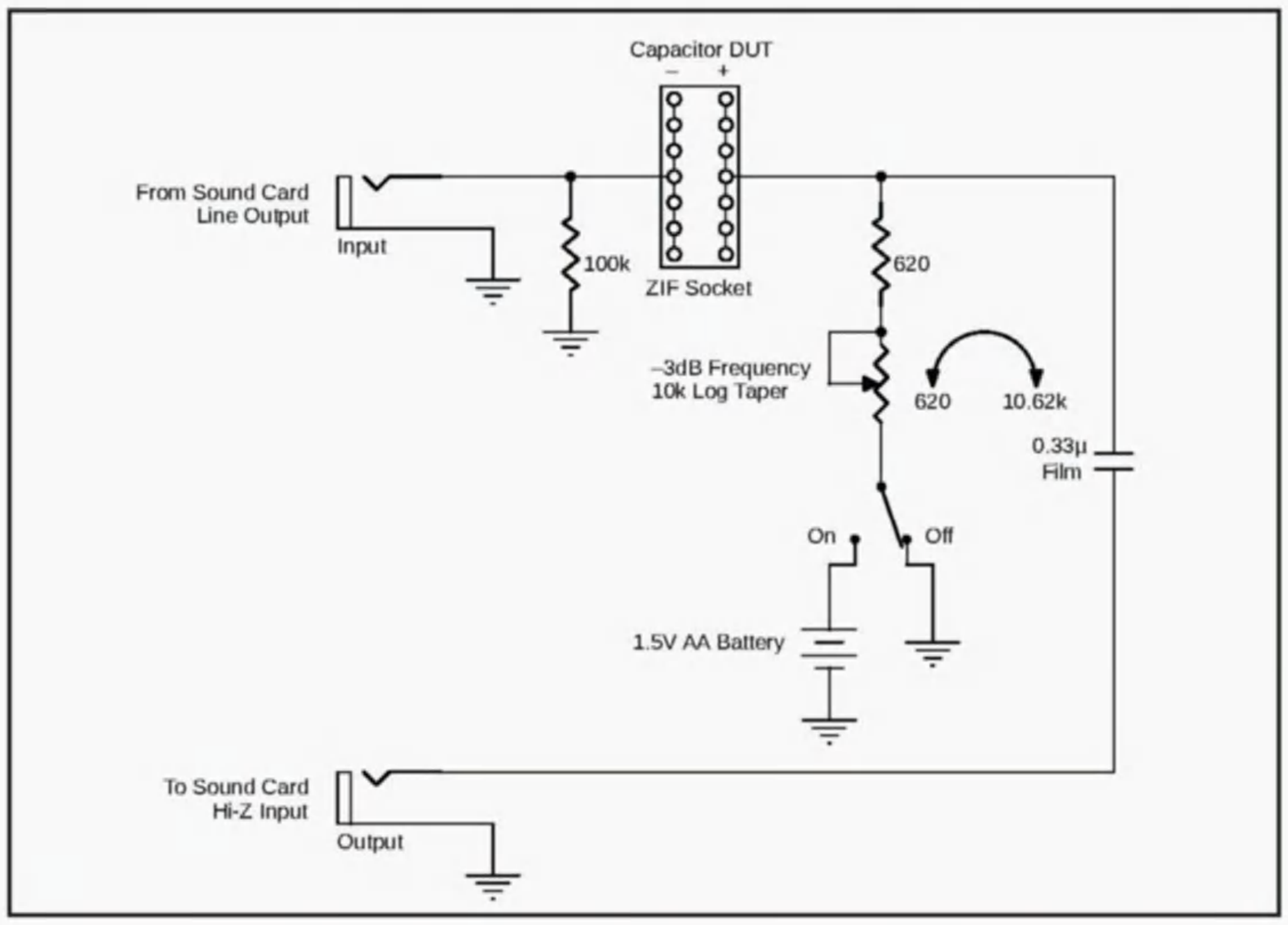


Figure 1: As you can see from the schematic, the tester itself is a simple device that relies on a high-quality sound card and suitable software to do the real work.

potentiometer setting the low end cut-off frequency. The On/Off switch adds the battery when needed to bias polarized capacitors, or grounds the resistors directly.

Figure 1 shows the tester schematic. **Photo 2** shows its internal wiring. As you can see, it's a very simple device! In use, the sound card's Line level output goes to the tester's input, and the tester's output goes back into the sound card's Instrument level input. My sound card's line output has a very low impedance and can drive a load less than 600 Ω without increasing its own already-low distortion. I actually tested this using a 1 kΩ trim pot. The distortion stayed at the same low value to below 500 Ω, so 620 Ω is a safe lower limit.

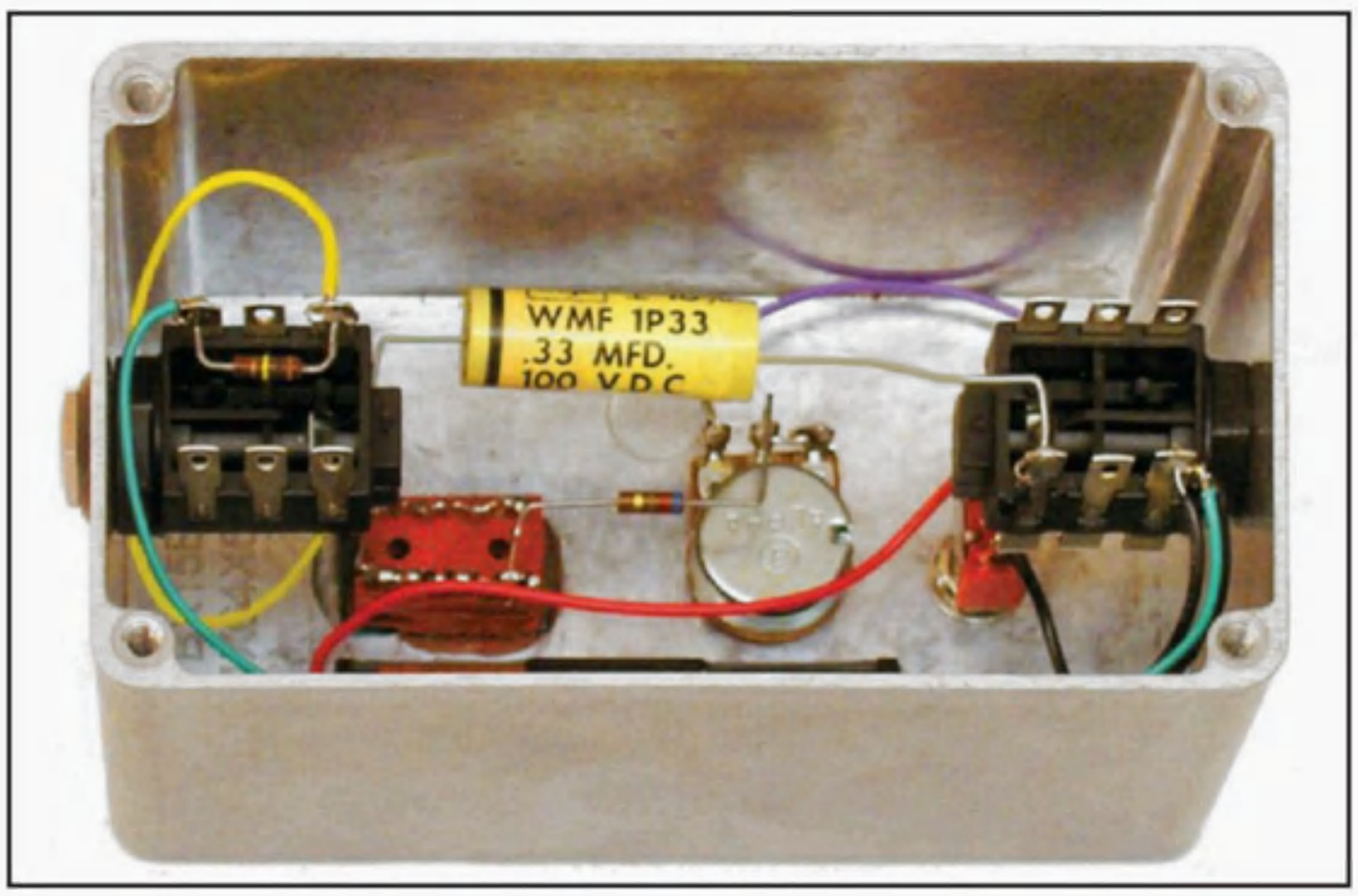


Photo 2: This is the tester's innards, showing the point-to-point wiring. No circuit board needed.

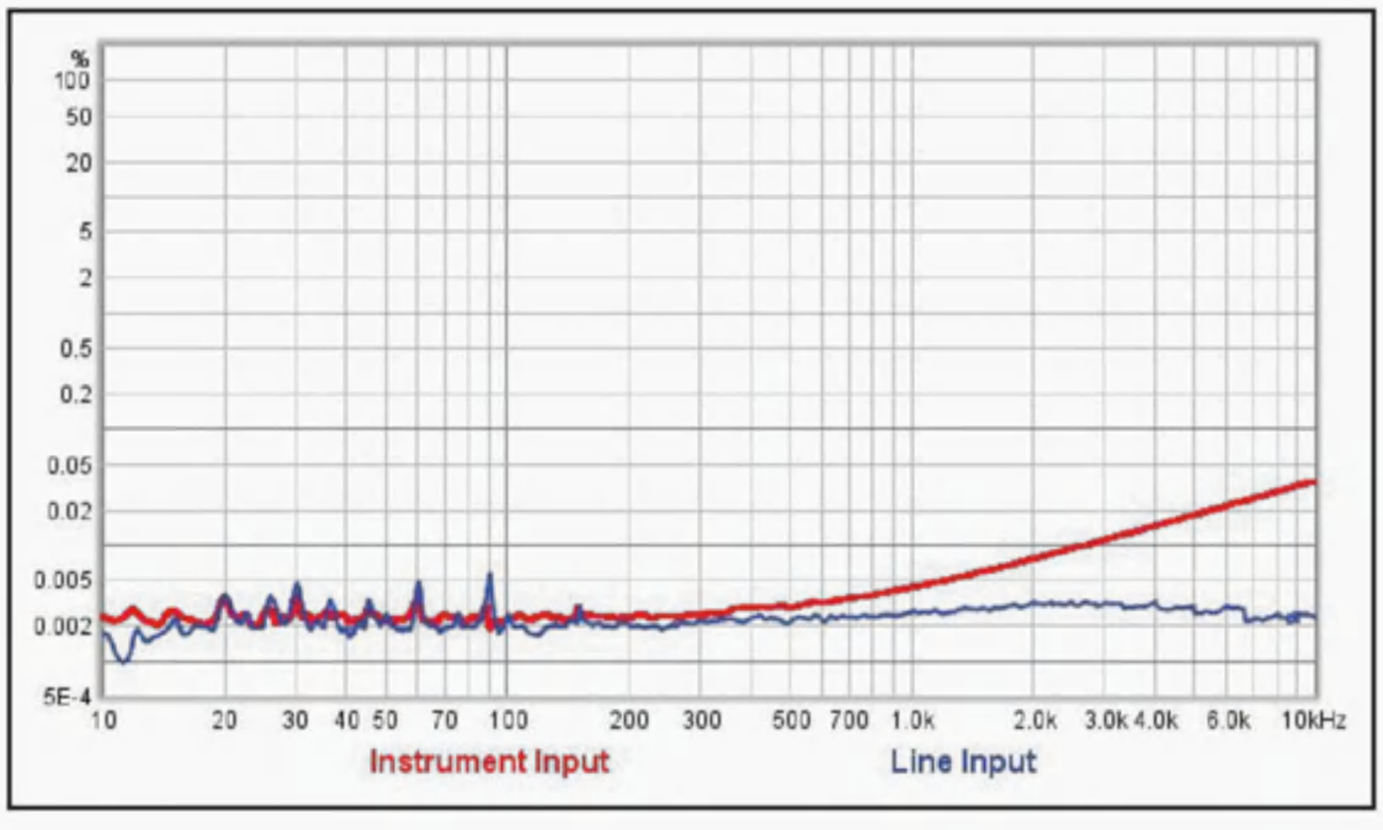


Figure 2: Connecting the sound card's output directly back to its own input allows REW to measure distortion vs. frequency of the sound card itself.

The inputs to my Focusrite sound card can be switched between line level with a claimed input impedance greater than 10 kΩ (I measured it as 24 kΩ), or set for "Instruments" that output a lower signal level and require a much higher input impedance. Passive electric guitars and basses need to drive an input that's at least 1M, and I measured this input at 2.9M, which is very high. The tester needs its own output coupling capacitor to block the 1.5 VDC when the battery is engaged. I measured the distortion of several high-quality film capacitors I already had in my parts bin, and none increased the sound card's own low distortion. So I chose the 0.33 µF Cornell Dubilier WMF Polyester Film capacitor shown in Photo 2 because it's old school and looks cool.

I used a 10 kΩ audio taper potentiometer to vary the load impedance, and the schematic shows the expected range of values from 620 Ω through 10.62 kΩ. But when I calibrated the panel labeling by measuring the resistance at each line marking, I discovered that the potentiometer is actually 9.33 kΩ. So the panel labels are correct. The load impedance potentiometer allows keeping the roll-off frequency within range to see distortion above and below that frequency. I used the terrific Electronics Assistant freeware calculator from the Electronics2000 website to help determine the best values.

As I previously mentioned, the tester includes a 1.5 V AA battery when needed to bias polarized capacitors to ensure that their plus terminal is always more positive than the minus side. Without this bias, the capacitor would add more distortion than normal. I didn't apply this voltage when testing the ceramic and film capacitors, though I tried that earlier and it didn't make any difference.

I also measured the distortion of a large value film capacitor driving the worst case 620 Ω load with and without the battery, and the response and distortion were identical. So that proves the battery presents a suitably low impedance at audible frequencies. However, this applies only for a fresh battery—as a battery drains its impedance rises. Since minimal current is needed to charge the relatively small capacitors this device tests, the battery in my tester should remain usable for years.

Measuring Distortion

Before I could measure the distortion of various capacitor types and values, I needed to confirm my sound card's own distortion. This is easy to do using a "loop-back" connection whereby the sound card's output is connected back to its own input. **Figure 2** shows the distortion measured for a direct connection using both

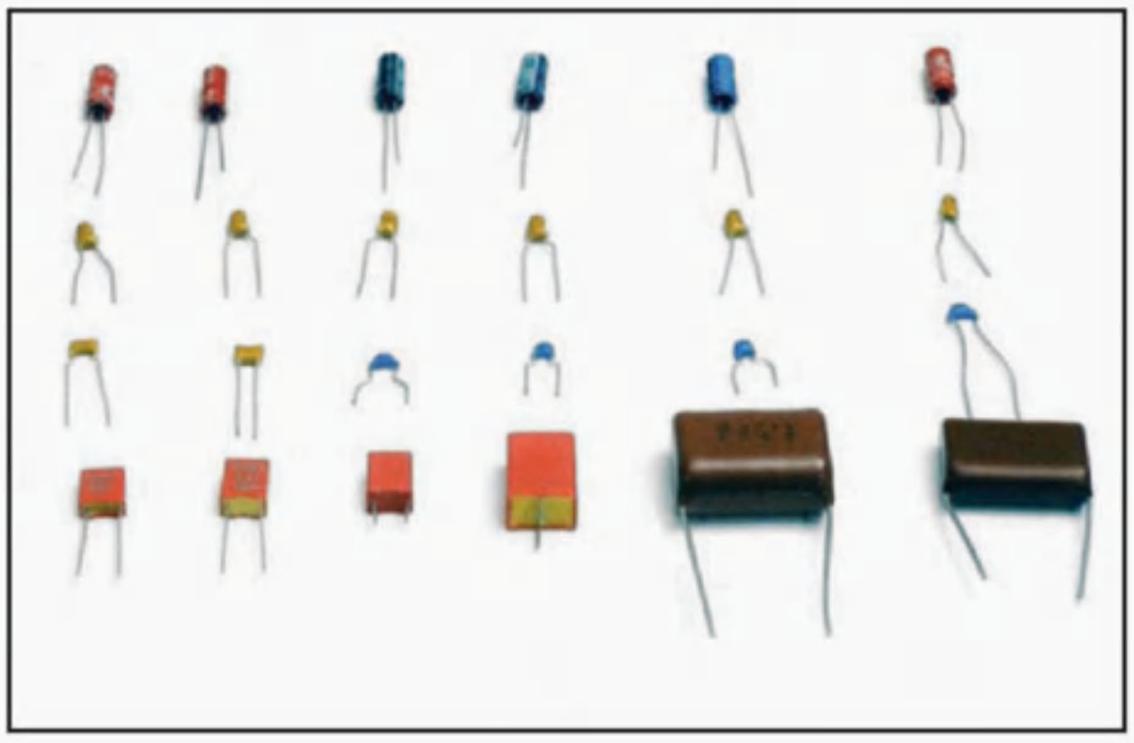


Photo 3: I measured all 24 of these capacitors for this article.

inputs. Focusrite specs this sound card at 0.001% total harmonic distortion plus noise (THD+N) for the outputs, and 0.004% THD+N at 1 kHz for the Instrument input I used. As you can see, the distortion is much lower at mid and high frequencies with the Line input than the Instrument input, and the Instrument input's distortion rises above 1 kHz.

Both measurements in Figure 2 were made with the sound card outputting –5 dBFS. Minimum gain for the Instrument input is 13 dB, so I set the same amount of preamp gain when I switched the input to line level. The capacitor distortion measurements that follow used the high impedance Instrument input to avoid interacting with my tester's 10 kΩ load impedance pot and 0.33 µF output capacitor. You can see little blips at 30 Hz multiples where the distortion at very low frequencies seems to increase, but that's most likely due to a small amount of hum being picked up.

For my tests I bought one capacitor each in six values of four types (see **Photo 3**). The four types I tested were film, aluminum "can" electrolytic, tantalum electrolytic, and ceramic. The six values tested were 0.22 µF, 0.47 µF, 1 µF, 2.2 µF, 4.7 µF, and 6.8 µF, though the figures that follow show only three of those values because all the values measured similarly. The entire set of 24 capacitors cost less than $10, though I feel sorry for the poor employee at Mouser Electronics who had to bag and label all of these single parts!

It wasn't practical to test very large value capacitors because that would put their low-frequency –3 dB cut-off at too low a frequency, and these tests need to show what happens well below that cut-off. To test a 100 µF capacitor at 100 Hz would require a load resistor of 16 Ω, which is much too low for my sound card to drive without adding distortion of its own. So I tested at 500 Hz for the smaller capacitors, and lowered the frequency for the larger value capacitors. Since I couldn't go below 620 Ω, tests for the larger value capacitors start rolling off at lower frequencies. But the frequencies are still high enough to see how the distortion rises below cut-off.

For these distortion tests REW was set to output a level of –5 dBFS through my sound card. This applied the maximum practical voltage across the capacitors, but without distorting the sound card's high impedance Instrument input that was set at its minimum gain. For my Focusrite 8i6 sound card, –5 dBFS equates to an output level of 0 dBu, or

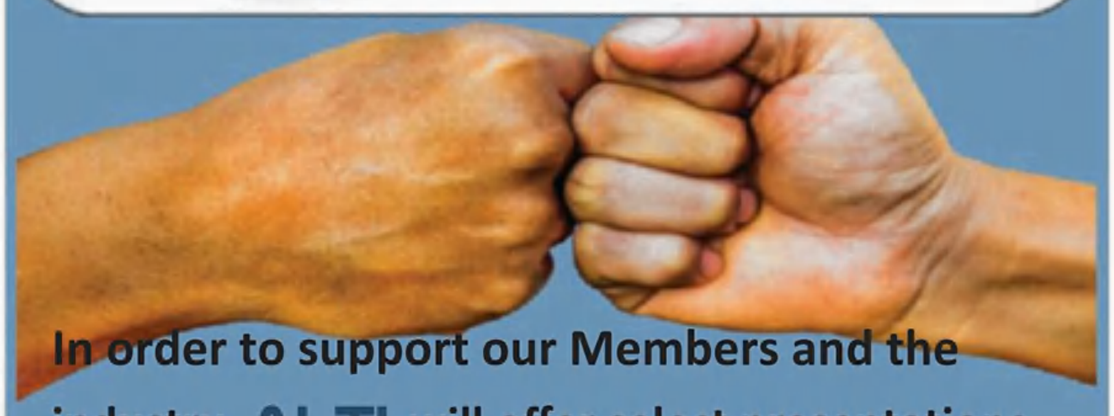
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0.775 V RMS (2.19 V_{PP}). **Figures 3–5** show the distortion of three values for all four capacitor types.

For clarity, the distortion comparisons in Figures 3–5 show only the total distortion amounts, but REW can also display the level of individual harmonics. **Figure 6** and **Figure 7** show the total distortion plus all of the individual harmonics up to the sixth for 4.7 µF film and ceramic capacitors. For the ceramic capacitor you can see that the third and fifth harmonics dominate, but for film types the second harmonic is the largest contributor. The numbers at the bottom are for the distortion amounts at 100 Hz. Since capacitor distortion varies with frequency, the numbers REW displays follow the current cursor location. So I put the cursor at 100 Hz when I exported these images.

To show the worst case for ceramic capacitors, the types I tested were either Z5U or X5R. Besides having high distortion, they're also microphonic due to using a piezoelectric dielectric that packs more capacitance into a very small physical size. NP0 and C0G ceramic capacitors are much better, but still nothing like film or polystyrene capacitors. For the piezo tests, I connected one each of the four capacitor types to the input of my Fender SideKick guitar amplifier. Guitar amps have a very high input impedance, which is needed when receiving audio from a piezo device (e.g., a

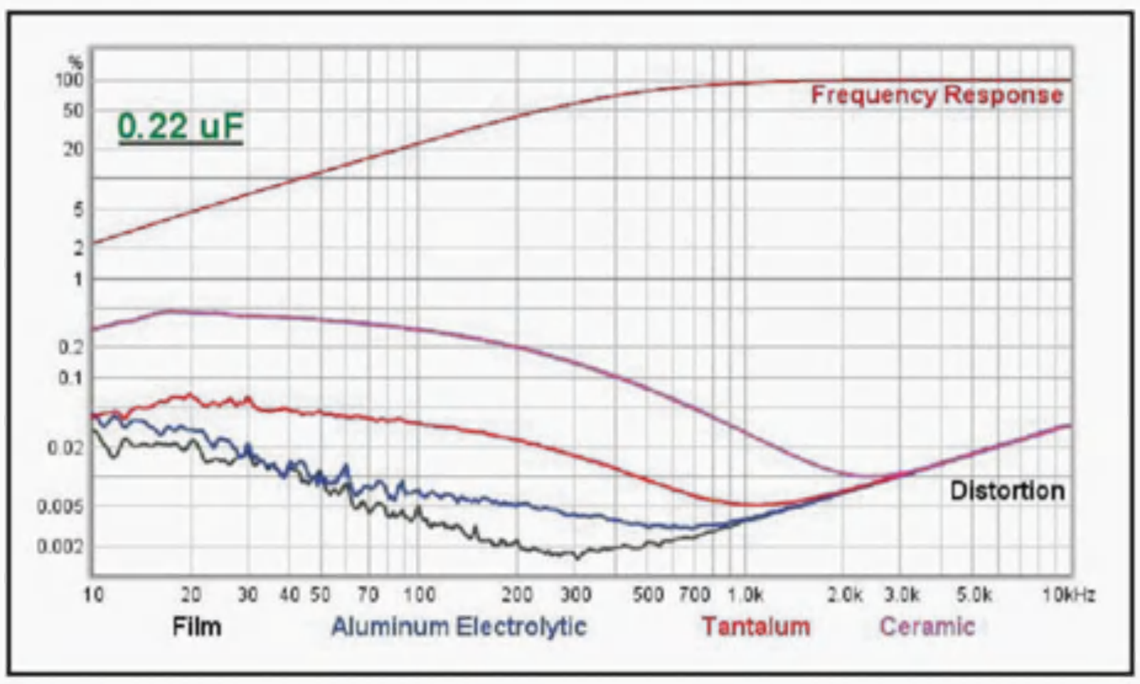


Figure 3: Distortion with a 0.22 µF capacitor

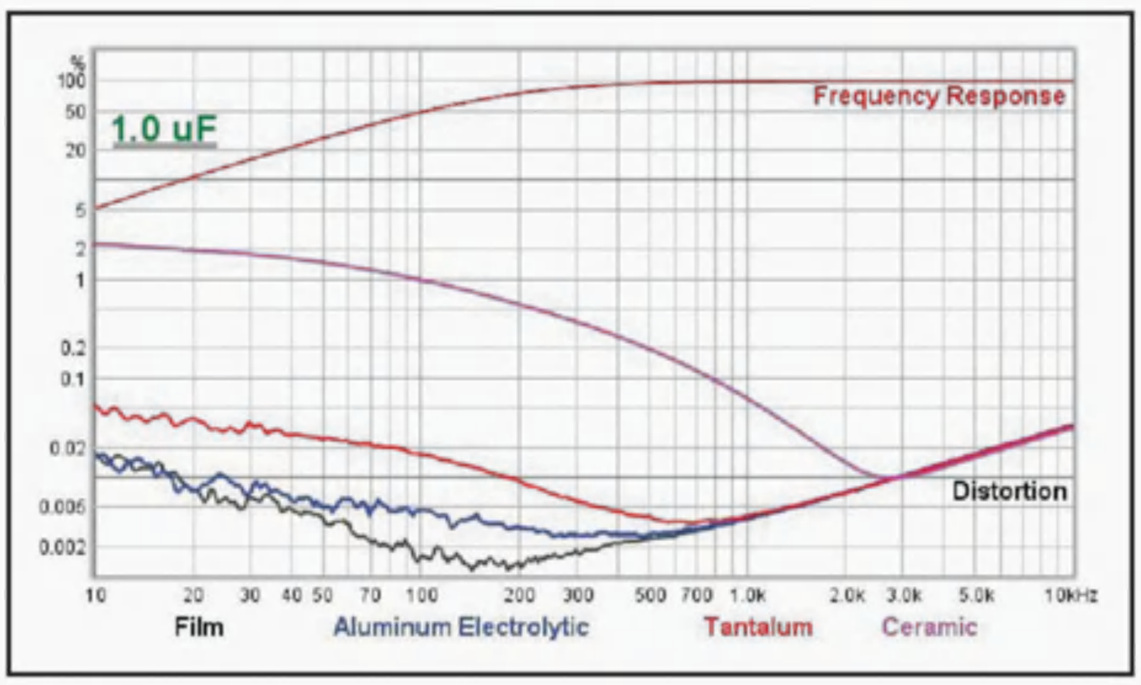


Figure 4: Distortion with a 1 µF capacitor

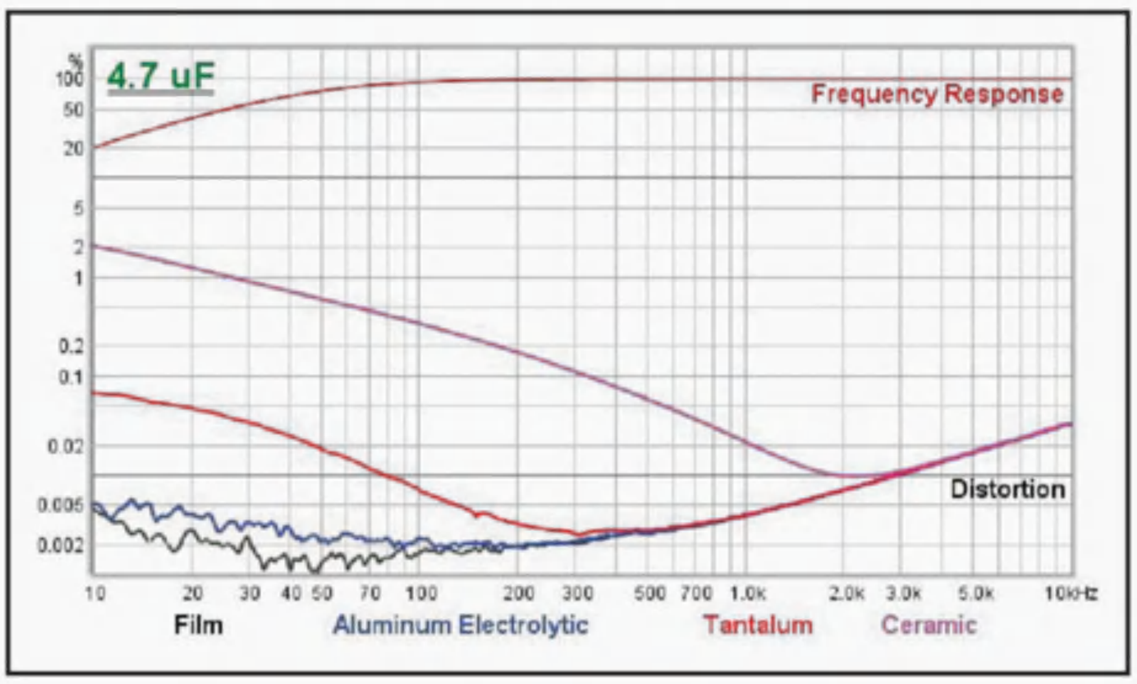


Figure 5: Distortion with a 4.7 µF capacitor

Figures 3-5: These graphs show the distortion of four types of capacitors for three values, with the tester's selected frequency response at the top. The rise in distortion at frequencies above 300 Hz is inherent in the sound card's Instrument input, and not a property of the capacitors being tested.

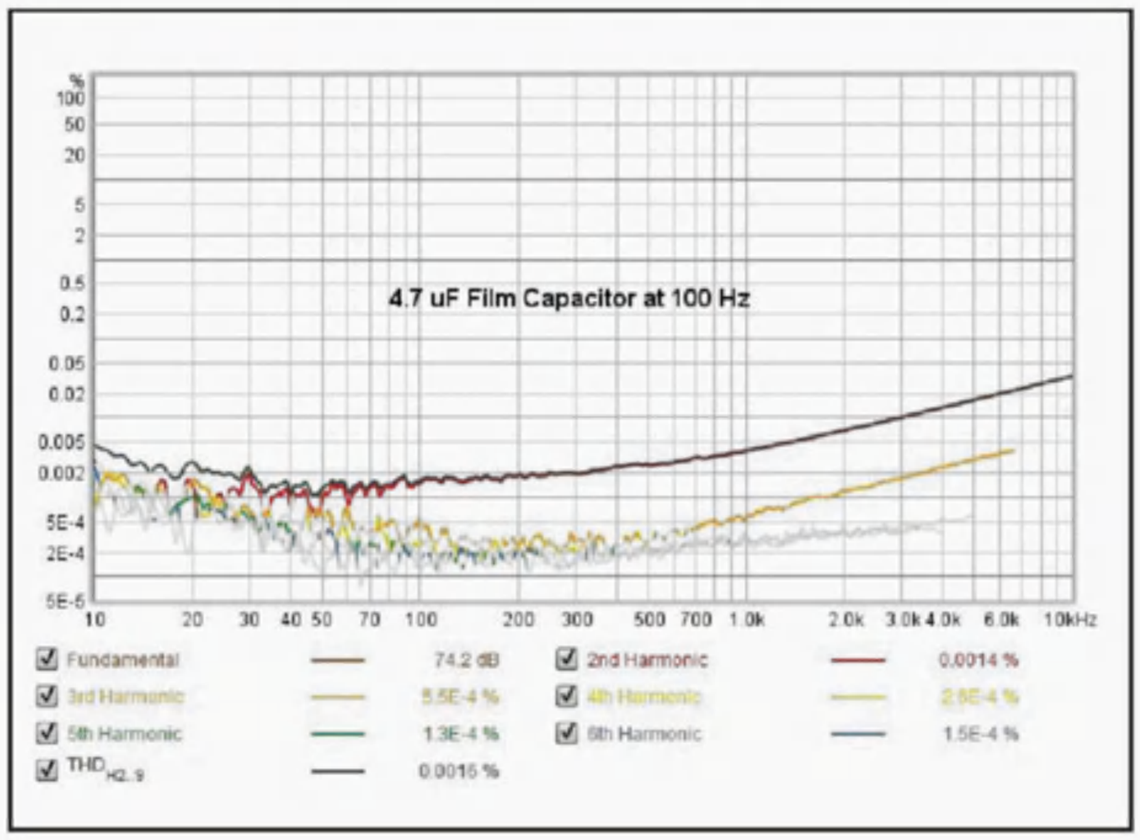


Figure 6: Besides showing the total distortion vs. frequency, REW can also display the level of each individual harmonic component. This graph shows the film harmonics.

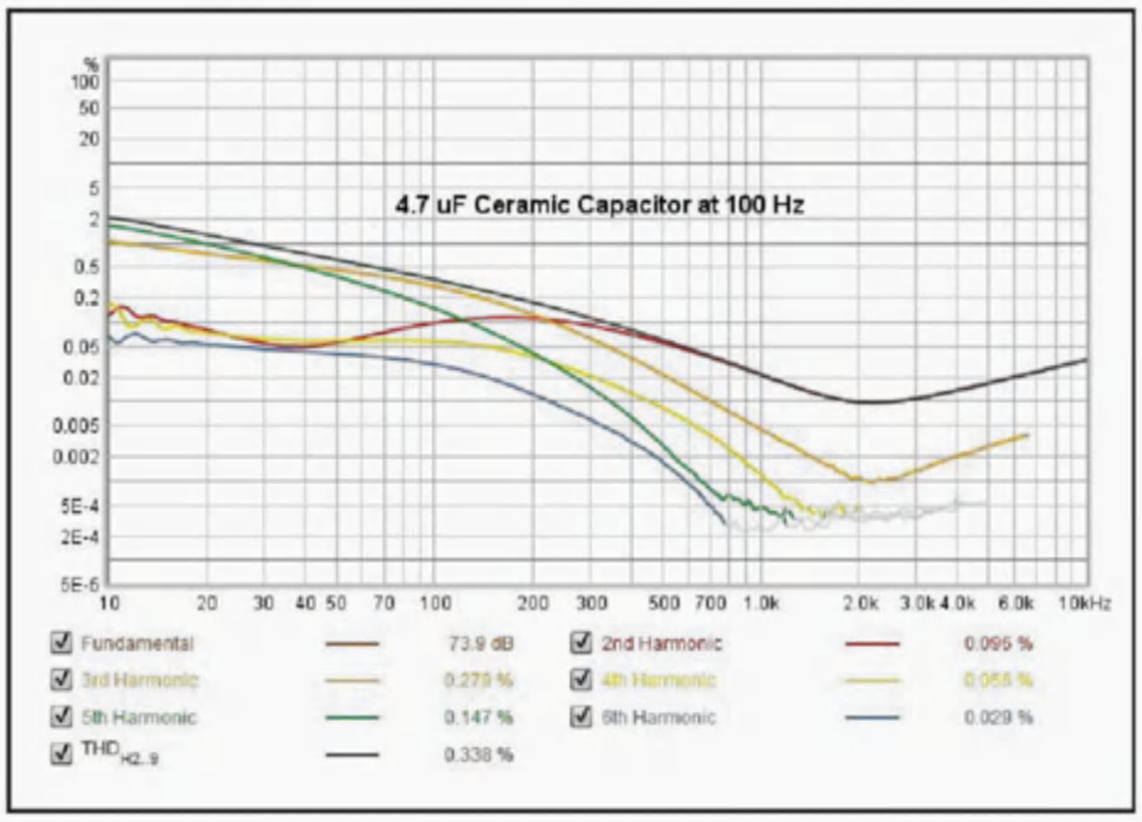


Figure 7:This graph shows the ceramic harmonics.

contact microphone). I put my Zoom H2 portable recorder 6" in front of the amplifier's speaker and cranked all three volume and gain knobs fully. Only the ceramic capacitor made noise when I tapped it with a small metal pliers. You can download the audio file listed in our Project Files on the *audioXpress* website to hear this noise, though there's a lot of amplifier hiss since the volume was raised to maximum.

As stated at the beginning of this article, my goal was to determine whether or not distortion is a problem for coupling capacitors at audible frequencies above 20 Hz. Looking at the measurements in Figures 3–5, it's clear that for all of the capacitor types other than ceramic, distortion is indeed a problem only at frequencies near and below their low-frequency roll-off. For higher frequencies even inexpensive aluminum electrolytic capacitors have distortion close to the low residual distortion of the sound card itself. Though it's equally clear from my tests that cheap ceramic capacitors should never be used in any audio signal path!

While preparing this article, I learned that it's very difficult to measure the tiny amounts of distortion in modern sound cards and film capacitors. I had to use a longer sweep time and multiple sweep repetitions in REW to increase the resolution over the program's default settings for measuring rooms. To get the highest resolution possible in REW takes four minutes for each sweep to complete! But even when a capacitor has slightly more distortion than the sound card itself, in practice anything less than 0.1% is mostly benign since at that point the distortion artifacts are 60 dB below the music. Of course, many audio devices employ more than one coupling capacitor, and all distortion accumulates. So it's best if each stage has even less distortion than is audible.



Photo 4: This Behringer HD400 "Hum Destroyer" contains two miniature line-level transformers—one for each stereo channel—to eliminate hum and buzz caused by ground loops between devices in an audio system. (Photo courtesy of Behringer)

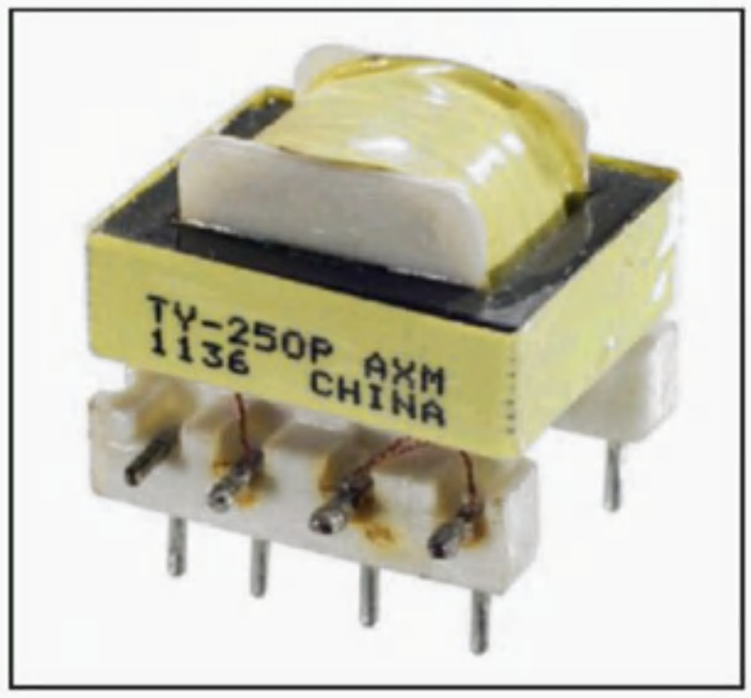


Photo 5: The Triad TY-250P is a popular audio transformer because it performs reasonably well and is very affordable. (Photo courtesy of Triad)

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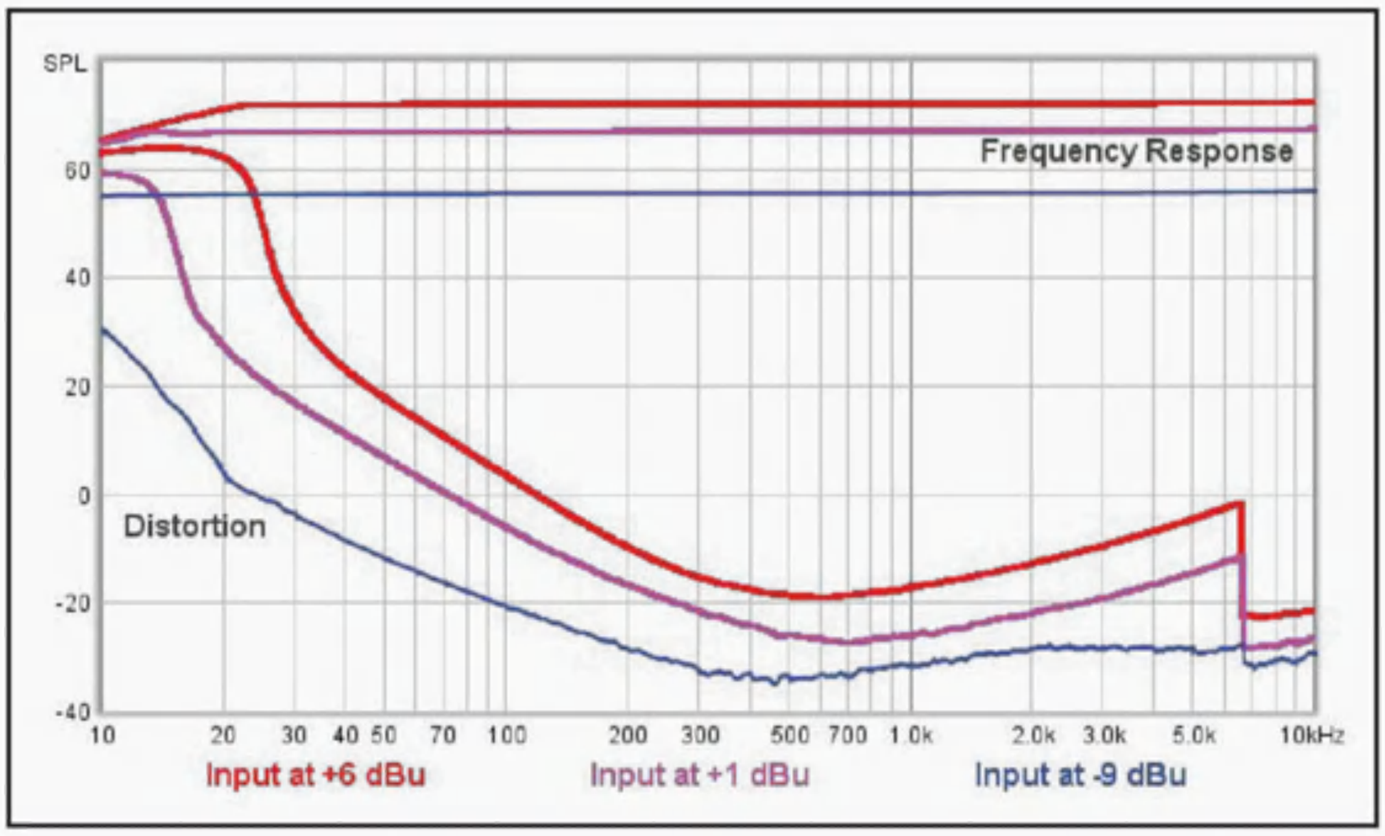


Figure 8: This shows the Behringer HD400 transformer's frequency response and distortion at three different input levels.

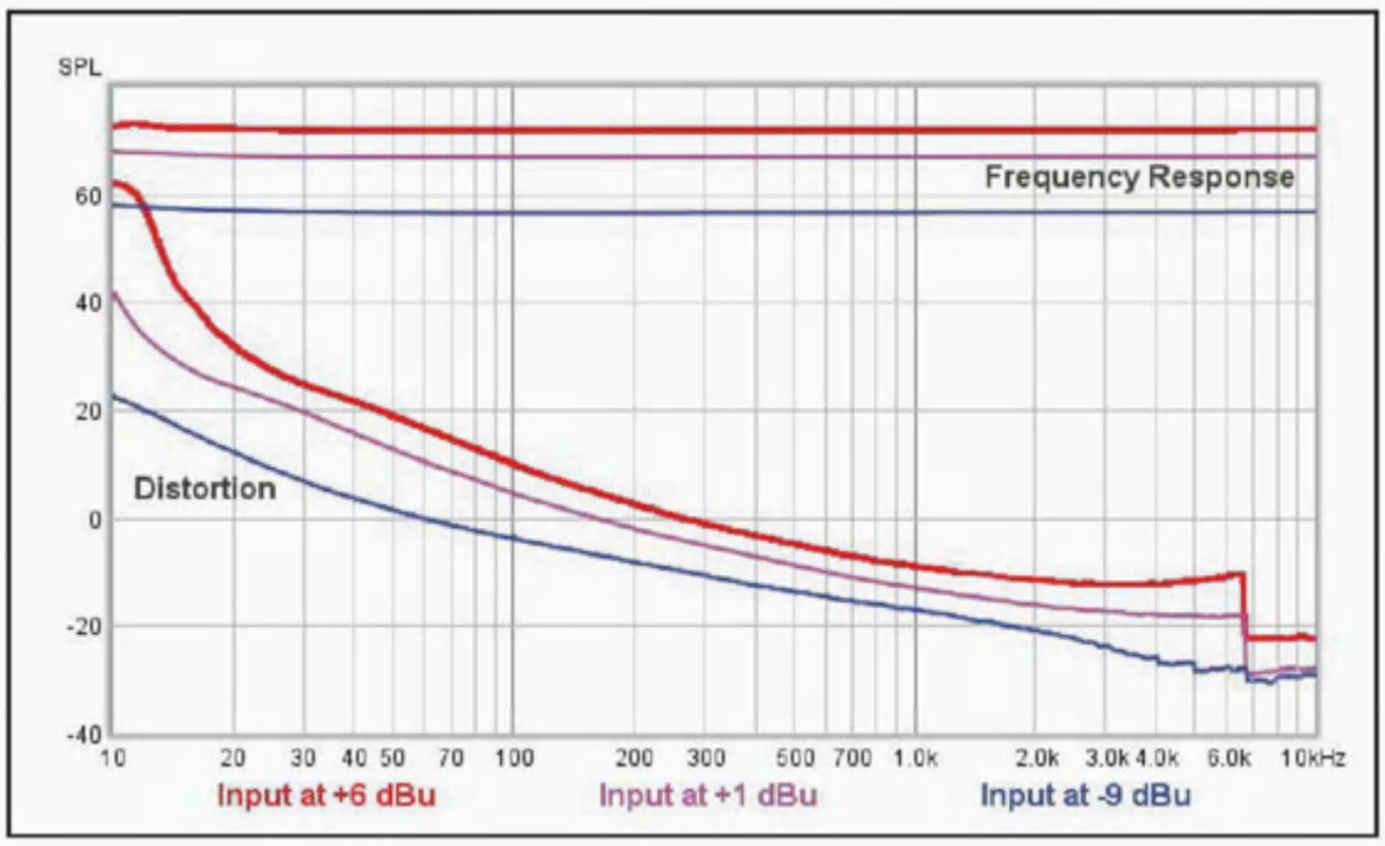


Figure 9: The Triad TY-250P's performance is similar to the Behringer.



Photo 6: The author built this parametric equalizer in the 1970s, and an explanation of the circuit, including the schematic, is available on his website.

About the Author

Ethan Winer has been an audio engineer and professional musician for more than 50 years. His Cello Rondo music video has received nearly 2 million views on YouTube and other websites, and his book *The Audio Expert* published by Focal Press, now in its second edition, is available at amazon.com and on Winer's own website ethanwiner.com. Winer is also a principal at RealTraps (http://realtraps.com), a US manufacturer of high-quality acoustic treatment.

Beyond Capacitors

REW can also measure the distortion of transformers and even entire circuits, without needing my capacitor test device. So for fun I measured two transformers I already had—the Behringer HD400 shown in **Photo 4** and the bare (uncased) Triad TY-250P that is shown in **Photo 5** and costs about $6. Unlike the capacitor tests that required driving the high impedance Instrument input of my sound card, I used the Line level input for the transformer and equalizer measurements that follow.

I measured the HD400 at several signal levels, and the results in **Figure 8** show differences not only in distortion but also frequency response at low frequencies. In order to show both frequency response and distortion for all three levels on the same graph, I set the vertical scale at left to display dB SPL instead of distortion percentages as in Figures 3–5. Every 20 dB level difference is a factor of 10, so the distortion is the difference in decibels between the Frequency Response line at top and the Distortion line below. For example, at 30 Hz the distortion of the largest 0 dBu signal spans two 20 dB marker lines, and so is about 1%.

I tested the HD400 transformer as it's intended to be used: fed from the low impedance balanced Line output of my sound card, then sent back into the 24 kΩ Line input. Since I used the balanced output from my sound card having both plus and minus signals, REW's –5 dBFS output became 6 dB higher into the transformer than when I used the unbalanced (single conductor) output for the capacitor tests. The actual levels are shown as dBu in Figure 8.

I tested the Triad TY-250P, shown in Photo 5, in exactly the same way as the Behringer, from the balanced output of my sound card at the same three voltage levels. The results in **Figure 9** show that this transformer is similar to the Behringer, but with more distortion at low frequencies and less at high frequencies. Note that for both transformers, the dominant distortion component is the third harmonic. This is why the distortion drops off suddenly around 7 kHz, which is one-third the upper frequency limit of the 44.1 kHz sample rate I used for all my tests.

The distortion of these inexpensive transformers isn't terrible, and at the typically low consumer voltage levels they could be considered "high fidelity" above 20 Hz or 30 Hz, anyway. By contrast, a high-quality audio transformer such as the Jensen Iso-Max CI-1RR has much better specs, including distortion at 20 Hz less than 0.04% at levels much higher than I used to measure these cheap transformers. But at $130, it's a lot more expensive. With audio transformers, you get what you pay for!

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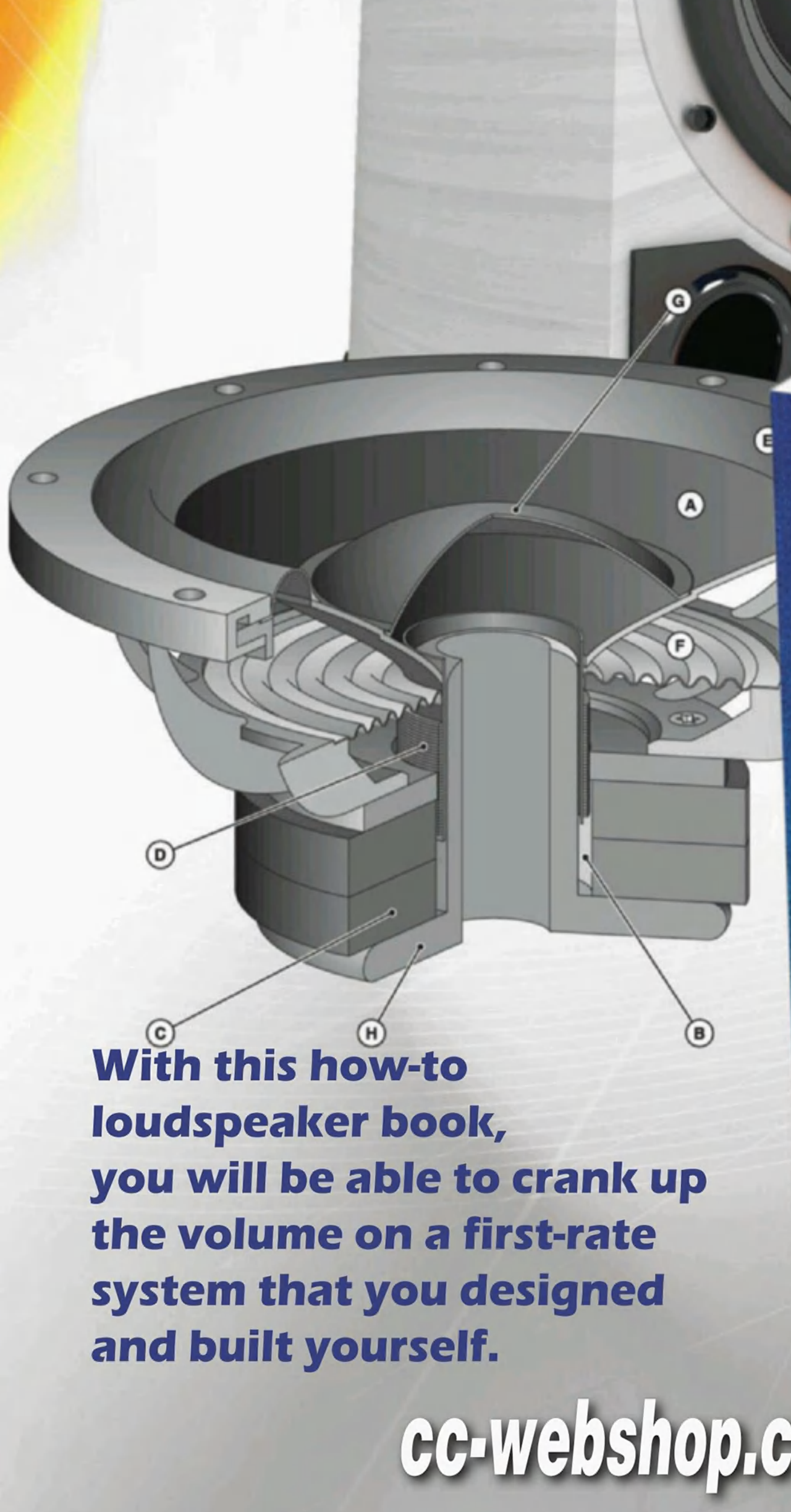




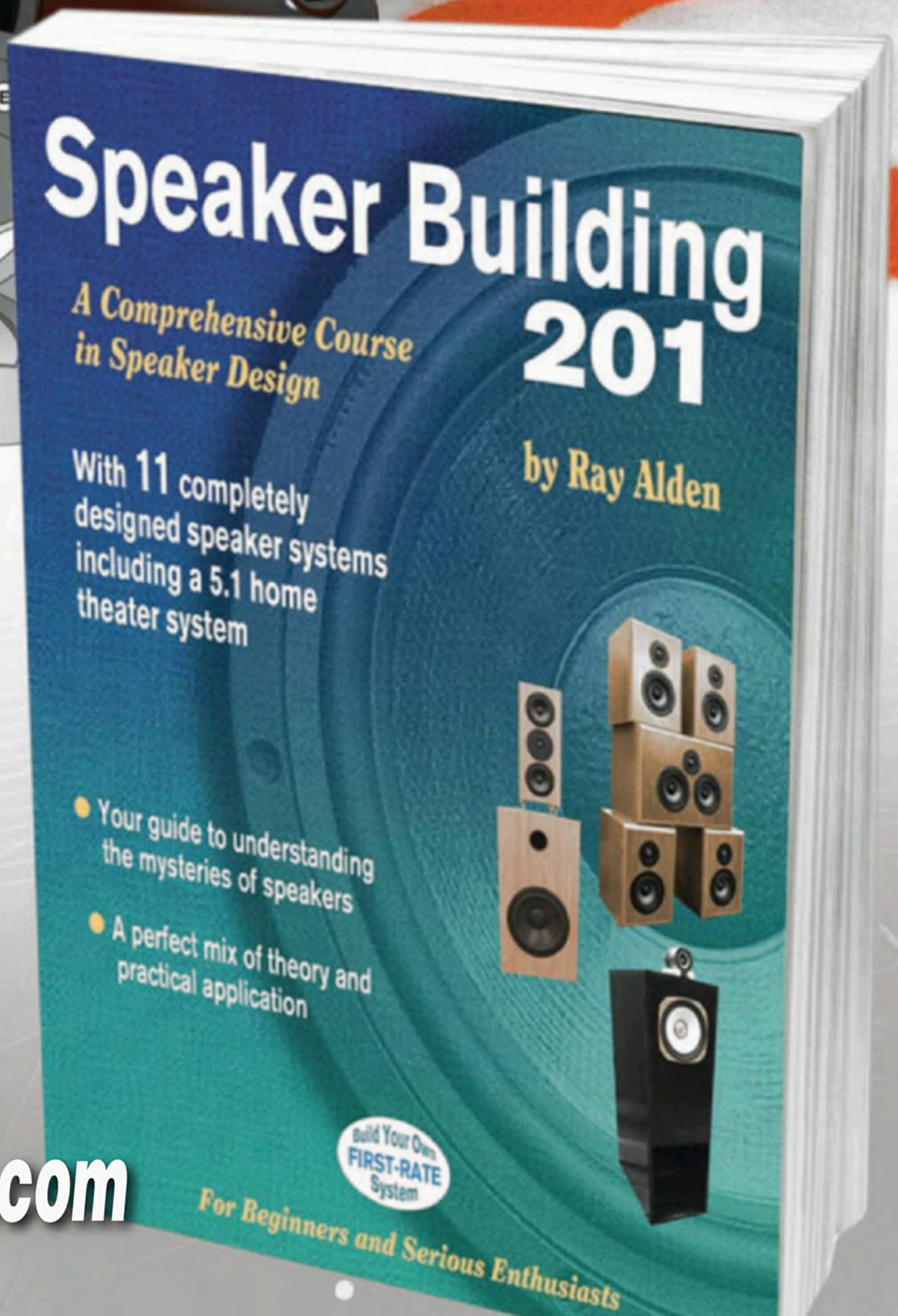
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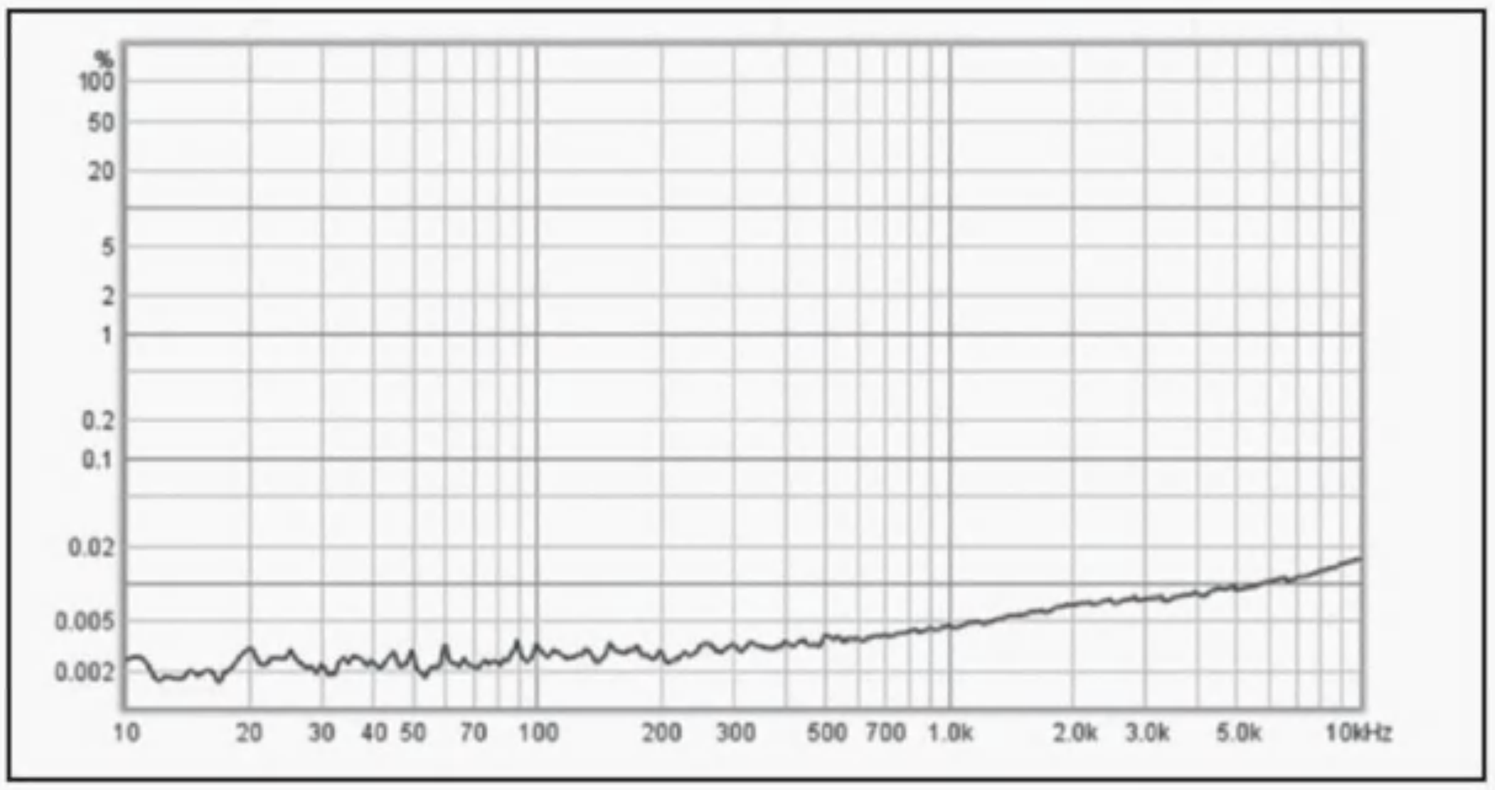


Figure 10: This classic parametric equalizer design adds little distortion, staying below 0.01% over most of the audible range.

Finally, I measured the distortion of the parametric equalizer in **Photo 6**. I built four of these in the 1970s for my professional recording studio, and an article describing the circuit along with the schematics are on my website. **Figure 10** shows the measured distortion—I was pleasantly surprised at how well this ancient unit performed in these tests!

The last point I'll mention is that REW doesn't measure intermodulation distortion (IMD), which is more damaging than the harmonic distortion I measured for this article. Where harmonic distortion adds musically related overtones to single frequencies, IMD creates sum and difference tones related to source frequency pairs. Worse, the added frequencies are not necessarily related to the musical notes that created them. So IMD is typically more audible and more objectionable than harmonic distortion. But the two distortion types go hand in hand, and they're always present together. So having low harmonic distortion also ensures low IMD.

Finally, I want to mention Doug Self's excellent book *Small Signal Audio Design*, now in its second edition, as the source of my interest in capacitor distortion. This book is a must-read for advanced audio hobbyists, and even professional circuit designers will find much of value. Another excellent resource that goes into great detail is Cyril Bateman's "Capacitor Sound" articles.

Project Files

To download additional the Capacitor Piezo test.mp3, visit

http://audioxpress.com/page/audioXpress-Supplementary-Material.html

Resources

C. Bateman articles, *Linear Audio*,
https://linearaudio.nl/cyril-batemans-capacitor-sound-articles

Electronics 2000, www.electronics2000.co.uk/download.php#assistant

Jensen Transformers, www.jensen-transformers.com/product/ci-1rr

RightMark Audio Analyzer, http://audio.rightmark.org/products/rmaa.shtml

Room EQ Wizard, www.roomeqwizard.com

D. Self, "Small Signal Audio Design," 2nd Edition, Focal Press, August 2014,
www.amazon.com/Small-Signal-Audio-Design-Douglas/dp/0415709733

E. Winer, "Spectrum Analyzer and Equalizer Designs,"
http://ethanwiner.com/spectrum.html

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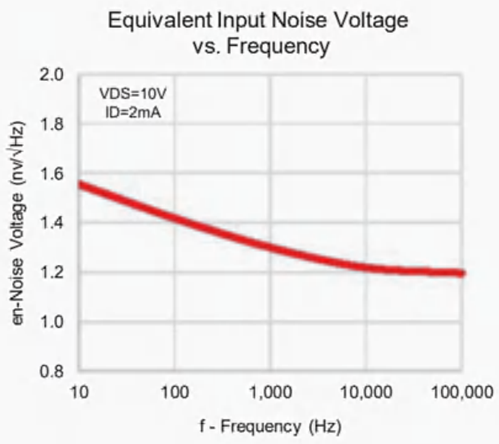
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Noise Analysis for FDA-Based Audio ADC Drivers

By
Michael Steffes
(United States)

In this article, Michael Steffes discusses the total output spot noise equation for the fully differential amplifier (FDA), which is an extension of the common op-amp noise expression. He also provides relative contributions to the total output noise power for the different terms, including 1/f noise effects to adjust the device noise for a wideband SNR calculation.

Starting from the previous THS4551 single to differential FDA solution driving a 24-bit audio ADC from a flat 2 Hz to 50 kHz measurement mic, simplify that circuit from an multiple feedback (MFB) filter to a simple gain stage shown in **Figure 1**. In this case, the original 2016 TINA model more closely matches the expected output spot noise than the updated 2019 model. The original model produces the measured 3.3 nV/√Hz input differential spot noise while the 2019 update reduced that noise to 1.7 nV/√Hz in error. This simplified circuit will be used first as a rough analysis prior to simulating the full output noise in the MFB filter.

To assess this (or any other) FDA stage for output noise use the relatively simple Equation 1 for total output spot noise (see Section 9.1.1 of Reference 1). This expression is forming the root sum of squared noise contributions. Each noise source is added as a power at the output then square rooted to get back to a total differential spot noise. For now, consider the FDA terms as only the flatband values.

$$e_O = \sqrt{(e_{ni}NG)^2 + 2(i_n R_f)^2 + 2(4kTR_f NG)} \quad \text{EQ 1}$$

In all FDA noise analysis cases, reduce each side of the feedback networks to their simplest equivalent networks and include whatever source impedance might be driving the inputs. Assuming those are matched on each side (as they should be nominally), the total output spot noise equation reduces to Equation 1 where:

- The noise gain (NG) term is $1 + R_f/R_g$
- 4kT is 16.5e-21J at 25C

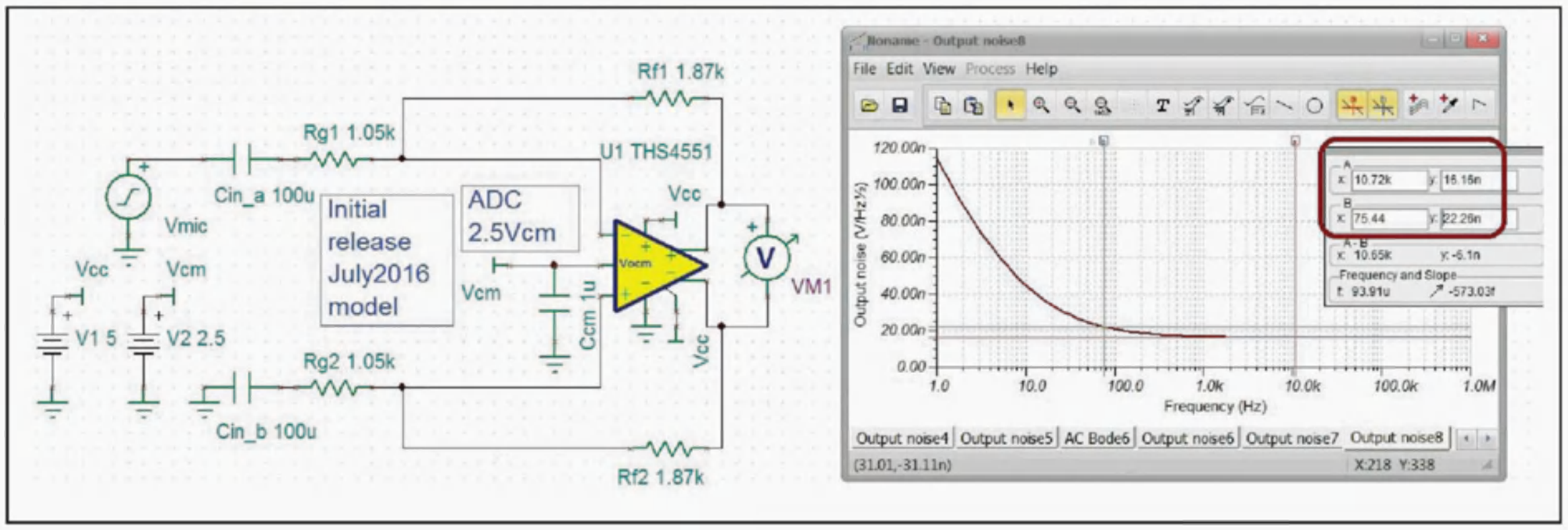


Figure 1: This graph shows the output spot noise for simplified gain only FDA single to differential stage.

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The Positivity of Tubes

This *audioXpress* Glass Audio Special Edition was a particularly difficult one due to the uncertainty of the circumstances related to the coronavirus COVID-19 outbreak. I've been working from home for many years already, but we have all had to adjust to the special circumstances—making sure family members are safe, organizing new daily routines, and trying to get on with our lives as much as possible. All with the sadness of knowing that this Glass Audio edition will not be handed out as usual at the High End Show in Munich, which was cancelled much like every other trade show and industry event around the world.

We have nevertheless doubled our efforts to stay in touch remotely with everyone in the *audioXpress* community and the audio industry. We wish everyone the best in getting through this ordeal.

As I wrote in our weekly newsletter *The Audio Voice*, I'm hopeful this will be temporary and I believe we should focus on planning for things getting back to normal. Meanwhile, we all need to keep doing our jobs, anticipate change, and remain confident and positive. Because, as with the virus pandemic, everything can change in just a few days.

At this time of the year, our team is usually also doubling up the efforts to create another edition of the annual *Loudspeaker Industry Sourcebook* (LIS), which given the unique global circumstances, is a more important tool than ever. There is no better way to find a supplier for anything audio-related, be it speakers, transducers, amplifier modules, components, or any other part for your projects—from prototype stages to high-volume production.

Unfortunately, in the past weeks we also received the sad news of the passing of a dear friend, *audioXpress* contributor and renowned tube expert, Gerhard Haas.

As with many of our longtime authors and contributors, very seldom do we have the chance to meet in person and share a good conversation. With Gerhard, at least I had the tradition of meeting every year at the High End show in Munich, the beautiful city where he lived, usually enjoying Bavarian beer and exchanging notes about the show. But most important, Gerhard would always share his ideas about the next monoblock tube amp designs on which he was working. And sure enough, every new project was described with enthusiasm as a total breakthrough.

audioXpress readers will be able to find Gerhard's projects dating back to the first years of the magazine, and even before that in the pages of *Glass Audio*, where his "Eintakt-A" Class A Single-Ended stereo amplifier was frequently featured in the advertising for the Texas-based International Audio Group.

I am certain that the owners of these simple but beautiful vacuum tube designs will continue to treasure them. Since they were usually made to order, or sold as a kit, many purchasers most probably had the chance to engage with Gerhard directly.

In this edition, we publish a modest tribute to his life and work, which is an excellent example of why the tube audio tradition for hi-fidelity and high-end audio applications remains so strong and vibrant even after many decades have passed since the heyday of tubes.

And the passion and dedication to tube circuits and tube DIY projects remains strong even among new generations. Every tube audio enthusiast I've met usually mentions a passion for tube circuits first and foremost because of the sound, but immediately after for the fact that to own and enjoy tube audio equipment represents in itself a continuous learning process. As Gerhard Haas often described it: a premium-class listening experience.

J. Martins
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Separate Output Noise Contributions for FDA Circuit					
	THS4551 FDA Input Noise Numbers Used				
	Flatband Value	**1/f Corner**	**Gain to Output**	**Total Output Power Contribution (nV)^2**	**% of Total Output Noise Power**
FDA Input Voltage Noise (nV/√Hz)	3.3	100 Hz	NG = 2.78 V/V	8.43E-17	32.7%
FDA Input Current Noise (pA/√Hz)	0.5	6000 Hz	R_f = 1.87 kΩ	1.77E-18	0.7%
R_f Johnson Noise (nV/√Hz)	5.6	0	1	6.16E-17	42.6%
R_g Johnson Noise (nV/√Hz)	4.2	0	R_f/R_g = 1.78 V/V	1.10E-16	23.9%
				Total (e_o^2)→	2.58E-16
				Total e_o=->	1.61E-08
				Total e_o (nV/√Hz)	16.05

Table 1: Total output noise power calculations for Figure 1

- e_{ni} is the intrinsic input differential spot noise voltage for the FDA being considered
- i_n is the intrinsic input spot current noise at each of the two FDA inputs

The last term in Equation 1 captures both the R_f and R_g noise terms where the "2" in front of that term and the current noise term reflects the two sides of the FDA adding noise power at the output. **Table 1** summarizes the elements of Equation 1 for the circuit shown in Figure 1 and their relative noise power contributions at the differential outputs. It is more instructive to compare the individual terms under the radical as percentages of total output noise power than spot noise contributions.

In Table 1, the separate R_g and R_f noise powers are shown, but these will combine to give the same result as using the last term in Equation 1. As is often the case, the resistor noise contributions dominate over the amplifier terms. The calculated total output spot noise of 16.1 nV/√Hz matches the simulation marker in the flat region in Figure 1.

The TINA model does include the 1/f noise region where the total output noise shows that corner (1.41 × flatband value) at 75 Hz. The FDA input voltage noise contribution (with a corner at 100 Hz in the model) is the only contributor to this with the other terms being flat moves the total corner down in frequency. These noise corner and flatband values in the THS4551 TINA model are a single "typical" device measurement with fit to the 1/f model. Actual parts vary slightly in their flatband values and considerably more on their low-frequency shape.

Comparing Designs and Devices

This discussion of a simple FDA circuit gives a good starting point for comparing designs and devices. So far, just the flatband noise combinations have been considered. With a bandlimited design concerned about SNR, including the effects of 1/f noise in the integrated noise would be very useful to

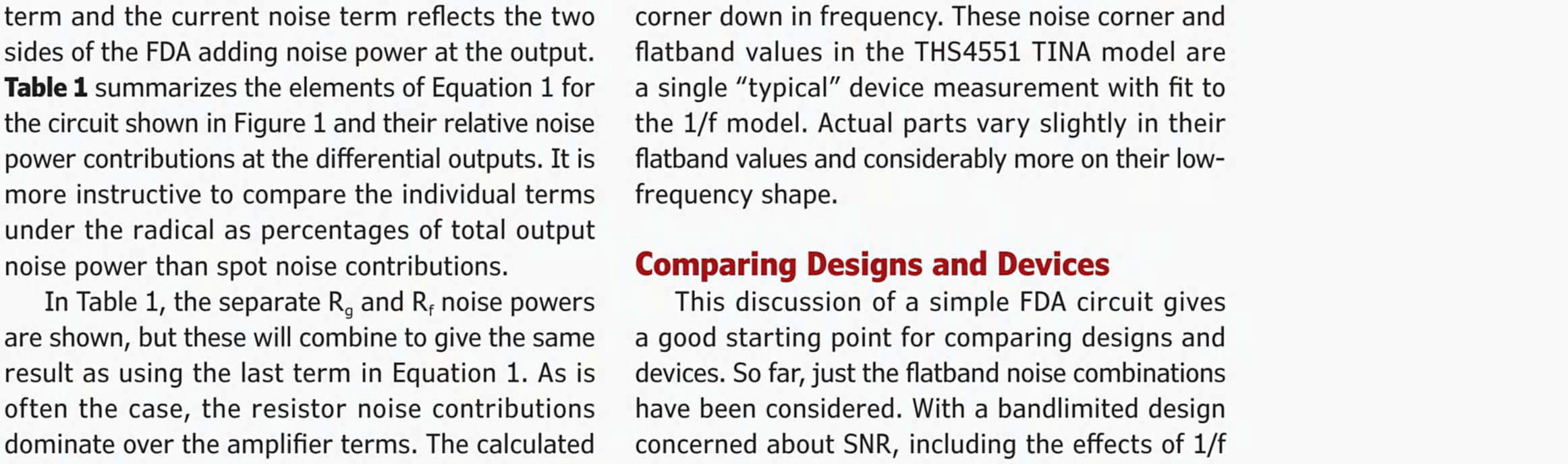


Figure 2: Here is the integrated output noise with a post RC filter for simple FDA circuit.

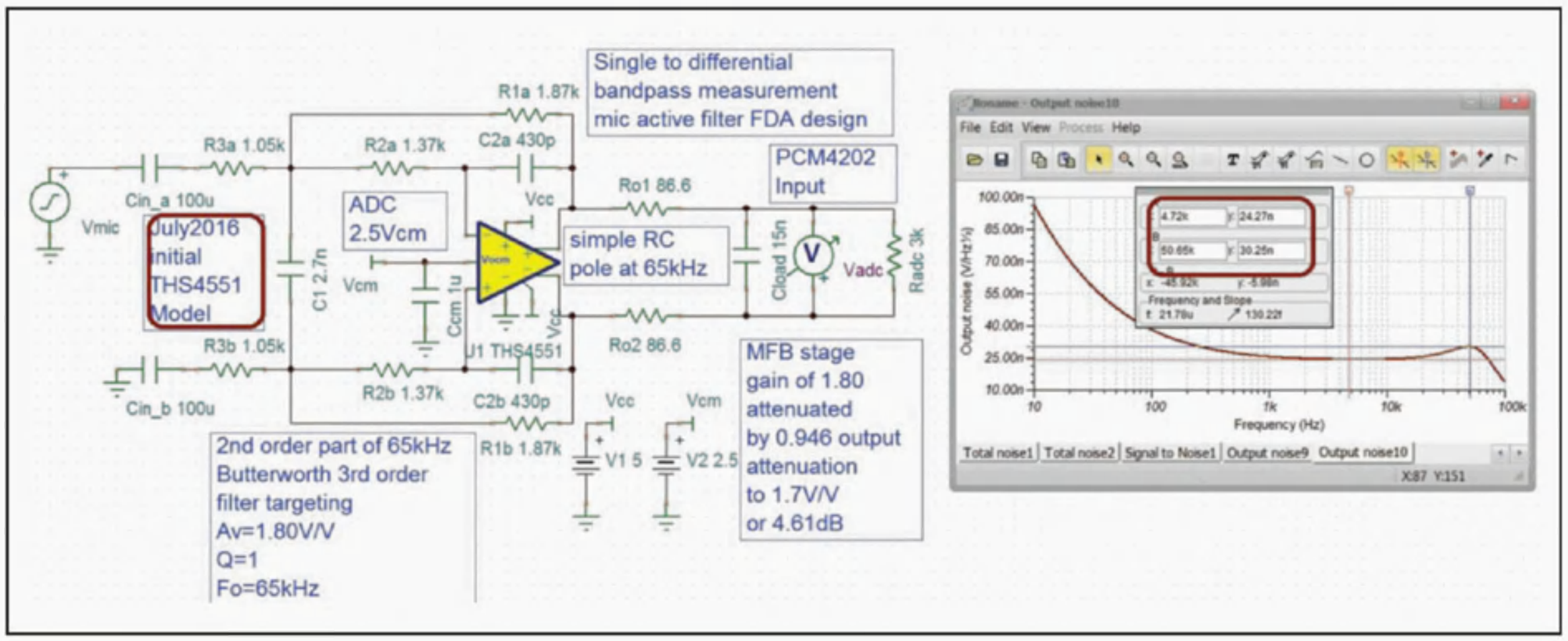


Figure 3: Here we show the output spot noise for full third-order 65 kHz Butterworth filter.

compare widely different 1/f noise corner devices. Getting to integrated noise (and then on to SNR) starts by defining a noise power bandwidth (NPBW). Briefly, this NPBW is the same brickwall bandwidth that will integrate to the same noise power as the actual output noise spectrum and response shape.

Going back to the third-order Butterworth filter at 65 kHz used previously, that last RC filter at 65 kHz gives a good place to extract an approximate NPBW. The noise spectrum at the output of the FDA will see that single pole roll-off in the response to the ADC. The simple conversion from F_{-3dB} to NPBW for a first-order pole is simply $(\pi/2) \times F_{-3dB}$. With a 65 kHz RC pole, that gives a 101 kHz NPBW.

One approach (see Section 6 in Reference 2) to estimating a full integrated output noise is to adjust the amplifier flatband numbers to include the 1/f effects in the NPBW integration bandwidth. That relatively simple adjustment is shown as Equation 2.

$$e_{eq} = e_n \sqrt{1 + \frac{F_{3dB}}{F_2 - F_1} \ln\left(\frac{F_2}{F_1}\right)} \qquad \text{EQ 2}$$

The lower F_1 and higher F_2 frequencies are the intended span of integration while the F_{3dB} is where the noise term has risen 3 dB from the flatband value going down in frequency (1/f frequency). For instance, setting F_1 =0.1 Hz and F_2 = 101 kHz for the input voltage noise term in Table 1 with an F_{3dB} = 100 Hz adjusts the flatband value very slightly up to 3.32 nV from 3.3 nV.

But imagine a much higher corner frequency FDA such as the MCP6D11 (see Figure 3-20 in Reference 3) with its already higher 5.1 nV/√Hz input voltage noise and 1.4 kHz F_{3dB} corner, where the adjustment given by Equation 2 yields an equivalent 5.4 nV/√Hz equivalent spot noise voltage for integration using the same F_1 and F_2 frequencies.

Making those slight adjustments to the calculations in Table 1 yields a 16.12 output spot noise. Then multiplying that by the square root of the 101 kHz NPBW will give an expected integrated noise of 5.1 μV_{RMS}.

Returning to the simulation circuit of Figure 1, adding the 65 kHz post RC filter, and running an integrated noise test from 0.1 Hz to 1 MHz gives the result shown in **Figure 2**.

This final value of 4.9 μV_{RMS} includes the 0.95 V/V attenuation from the FDA differential output noise to the ADC inputs in the RC filter. That implies a

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http://ww1.microchip.com/downloads/en/DeviceDoc/MCP6D11-Data-Sheet-DS20006162A.pdf

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http://www.ti.com/lit/an/sboa114/sboa114.pdf

5.2 μV_{RMS} at the FDA output pins where the RC filter R's are adding a small amount of noise over the expected 5.1 μV_{RMS} value.

TINA also offers an SNR feature where it will compare this swept integrated noise to a specified RMS signal level at the measurement point. The maximum input for the PCM4202 24-bit audio ADC is 2 V_{RMS} where running that SNR simulation gives 112 dB for the simple circuit of Figure 2. This agrees with a hand calculation of: SNR = 20 × log(2 V/4.9 μV) = 112 dB.

These approaches provide a useful screen of possible FDAs and R values to trade off cost, power, and noise performance. The noise can always be improved lowering the resistor values. That will increase the loading on both the source and FDA outputs both of which might reduce harmonic distortion performance.

Going back to the full third-order Butterworth filter shown previously, and running a spot noise at the ADC inputs in **Figure 3** shows the noise has increased and is certainly not flat over frequency. The spot noise in the flat region has increased to 24 nV/√Hz from the simple gain circuit 16 nV/√Hz shown in Figure 1. This is largely due to the added R2 resistors inside the resistor summing network required by the MFB filter design (see Appendix A in Reference 4).

There is also a noise peaking around the filter F_o frequency produced by the noise gain peaking inside the MFB filter design described previously. The simulation shown in Figure 3 has substituted the original 2016 THS4551 model to get the correct noise terms in the simulation.

Re-running this full design using a 2 V_{RMS} maximum output gives a 108dB SNR. Not as much of a degradation from 112dB partially due to the bandlimiting effects in the active filter for some of the noise terms above 65 kHz.

So far, this discussion has assumed a perfectly balanced feedback ratio network. When that becomes unbalanced (as all do at some level), the output common mode noise will leak into the differential noise terms. Normally a very small effect, it can become dominant with the much higher 1/f noise corners in the common mode (CM) noise spectrum. Including and reducing those effects will be considered in the next article.

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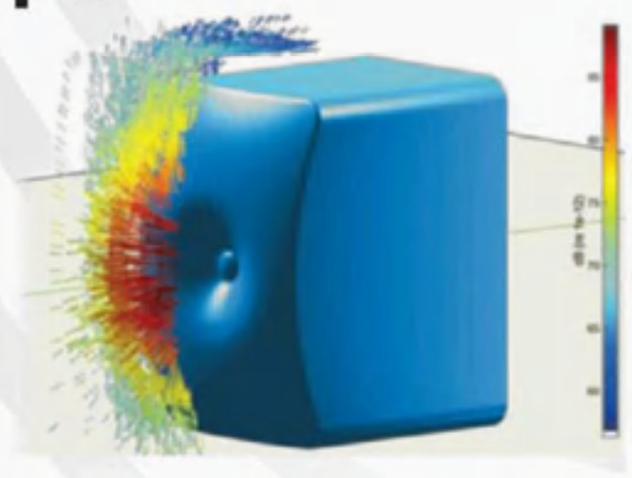
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You Can DIY!

Build This MoFo!

By
Mike Rothacher
(United States)



Mike Rothacher dreamed of a simple DIY amplifier project—a single-stage, single-ended, Class A, with high input impedance, low output impedance, and of course, he didn't want it to be too expensive. He also wanted a MOSFET Follower (MoFo), powered by a simple laptop adaptor. After waking up from his dream, Rothacher built it and shares all the details!

I couldn't sleep. I had designer's block. It was somewhere around 2:00 AM and I found myself digging through a pile of old capacitors at Electro-Bowl, an all-night electronics surplus and bowling alley. There aren't a lot of establishments where you can pick up a few transistors, then roll a few frames 24-7. I'd been there on more than a few sleepless nights. It kept me off the streets.

I could hear the rain outside tapping on the metal roof of the 14,000 ft^2 Quonset hut that was home to 50-odd years of curated electronic detritus and 10 well-worn but always freshly waxed lanes. The proprietor, Carl, sat behind the counter listening to his favorite late night AM radio station. The radio host was jabbering on about alien bases on the moon, or something like that.

Carl looked at me over his reading glasses and cleared his throat. "Ahem, are you workin' on another DIY article, kid?" Carl asked. I had been coming to Electro-Bowl for a really long time and Carl still called me "kid" even though I'm 50.

I'm trying to think of a cool project for Burning Amp," I said.

"Burning Amp? What's that?" he asked.

"It's like a comic book convention for audio nerds," I explained. "It's in San Francisco, CA."

He shook his head. "What'll they think of next? In my day, all we had was ham fests."

"I'm trying to come up with a fun amp project I can take to the show."

Carl pondered this for a moment. "You should talk to The Dude about your project. I bet he can help. He's here tonight."

"The Dude's here now?" I croaked. The Dude was legendary. Some say he'd been some kind of rocket scientist or particle physicist who left the science racket to pursue a monastic life of meditation and designing audio gear. He was widely known as an audio guru of the highest order, and a really good bowler.

"He's on lane 7," Carl said, glancing toward the far end of the store. He held out a pair of size elevens for me. "Shoe rental is on the house tonight. Good luck, kid."

I laced up my bowling shoes, turned, and walked slowly past the parts aisles, transformers, tubes, and semiconductors, then past lanes 1 through 6. A short distance away stood "The Dude" waiting for the pins to reset.

"Always check the date codes," he said taking notice of me, but not looking up.

"Excuse me?" I said.

He pointed to the box of power supply capacitors I had under my arm. "Carl never gets rid of anything, and he's had some of those caps since the 1970s, so you gotta check the codes."

"Oh, right Dude," I said. "Good advice."

He grinned. "So, what's up, man? I heard you need some help with your project for Burning Man."

"Well, it's Burning Amp actually, but yes, I'm kind of stuck."

"Yep. You got designer's block. That's cool, man. I can help out." "Ok, so what do you have in mind?" he asked as he shuffled silently up to the foul line and rolled." "Well, I'd like to do something really simple that beginners will like with really low parts count, and easy to solder-up and listen to in a few hours," I said. "Class A, of course. I'm thinking single-stage, single-ended, with high input impedance, and a nice low output impedance to drive my latest speaker project," I continued. "Low distortion, good bandwidth, not too expensive, and buildable with easy-to-find parts, nothing exotic."

The Dude grinned. "Wow, man, that's a pretty long list. I guess you want it to sound good too." The Dude thought for a minute, snapped his fingers, then, nodding he said, "No problem, man. You should make a MoFo!"

"I'm sorry, make a what?"

"A MoFo. You know, a MOSFET Follower. There's no voltage gain and you need a preamp that can swing all the voltage for your full output power, but it checks all the boxes. You get low-distortion, high input impedance, low output impedance, and good bandwidth since there's no Miller effect. It's single-stage, single-ended, and really, really simple." He tore-off a blank scoresheet, turned it over, sketched something on the back and handed it to me. "Here you go, man! It doesn't get much simpler than that."

Figure 1 shows the schematic that he gave me.

"Wow, that is simple," I said, "but what about the power supply?"

"Just use one of those laptop switchers," he said. "That'll be great for beginners. You could use anything decent from like 12 V to 24 V."

"Oh, great, but don't those have a lot of ripple?" I asked.

"Sure, but the really high drain resistance of the IRFP250, and the fairly low resistance of the inductor at the ripple frequency divide down the ripple voltage, so you won't get a lot of noise on the output," he said as he lined up for another roll.

I studied the schematic and fixated on L1, the inductor attached to the MOSFET's source. "Dude, won't the source degeneration raise the output impedance?"

He pointed to the schematic. "This is a Source Follower. You're taking the output from the source, not the drain, and the feedback is voltage-derived series-mixed, so you get effective curves that are more 'triodey' than a triode."

I wasn't entirely sure what The Dude meant by "triodey effective curves" but I decided to let that sink into my subconscious for a while. "So, why use an inductor anyway? Why not a current source or something?"

"Wow, man, I can see you like to worry about stuff," he said.

"Sorry, Dude. I guess it's an audiophile thing."

He took a seat on the bench across from me and adjusted his sunglasses. "Well, for convenience and safety, we're using laptop power supplies, and they work really great here, but you're limited to the lower voltages they offer, like 12 to 24 volts. So, if you want to get some real power out of this amp, you need a little trick to boost the efficiency. The inductor handles this without any special active circuitry because it can store energy in its magnetic field. The energy gets stored and released under the control of the gain device, and as a result, the amplifier can deliver almost twice the supply voltage to the load and the efficiency approaches 50%. Plus, we're able to use a smaller inductor in this circuit because the load's in parallel with the really low Source resistance of the MOSFET." Then he lowered his shades a little and looked me in the eye. "And, it's simpler."

"Oh, I get it. That's clever, Dude. So, will it sound good?"

"Well, you gotta build it and judge for yourself my friend, but I think you'll dig it." The Dude stood up. "Well, it looks like my bowling partners are here. Say hello to Al and Dick." Two guys walked up, their names were stitched on their bowling shirts, Einstein and Feynman.

I shook my head a few times in disbelief, then my vision went blurry. Just then, music began playing on the loudspeakers overhead. It was indistinguishable. There was singing, was it French? Then, over the music, loud steady beeping. "Dude, what's that beeping?" I asked.

The Dude grinned back at me. "Time to wake up, man. Now go build something."

My alarm clock wailed.

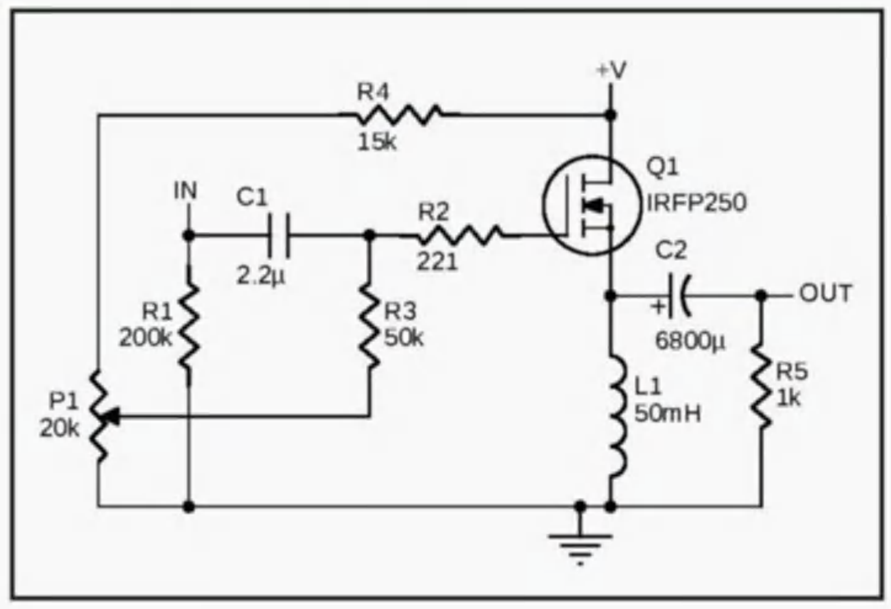


Figure 1: The schematic for the MOSFET Follower checks all the boxes. You get low-distortion, high input impedance, low output impedance, and good bandwidth.

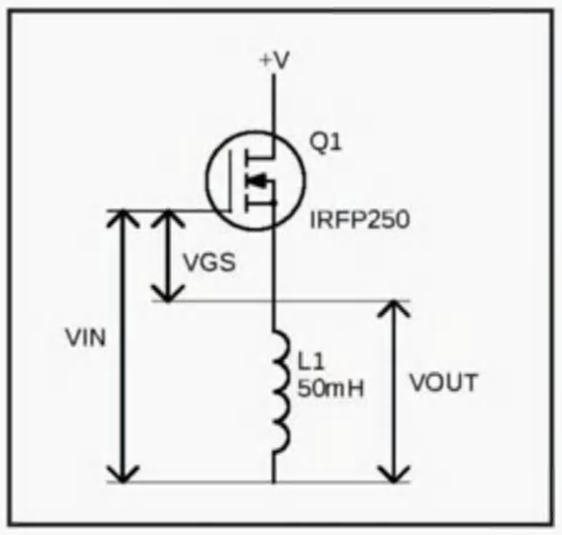


Figure 2: This schematic shows how the feedback works in the Source Follower.

I turned off the alarm and rubbed my eyes, took out some graph paper and sketched the schematic from my weird dream. It was a dream, right?

MoFo Mojo

Figure 2 shows how the feedback works in the Source Follower. The inductor loads the source and the output signal appears across the inductor. The Gate-Source signal is the difference between the input and output signals (VIN – VOUT = VGS).

Figure 3 shows a set of characteristic curves for an IRFP250 MOSFET. They sure look a lot like Pentode curves, so what was The Dude talking about when he said MoFo has "triodey" curves?

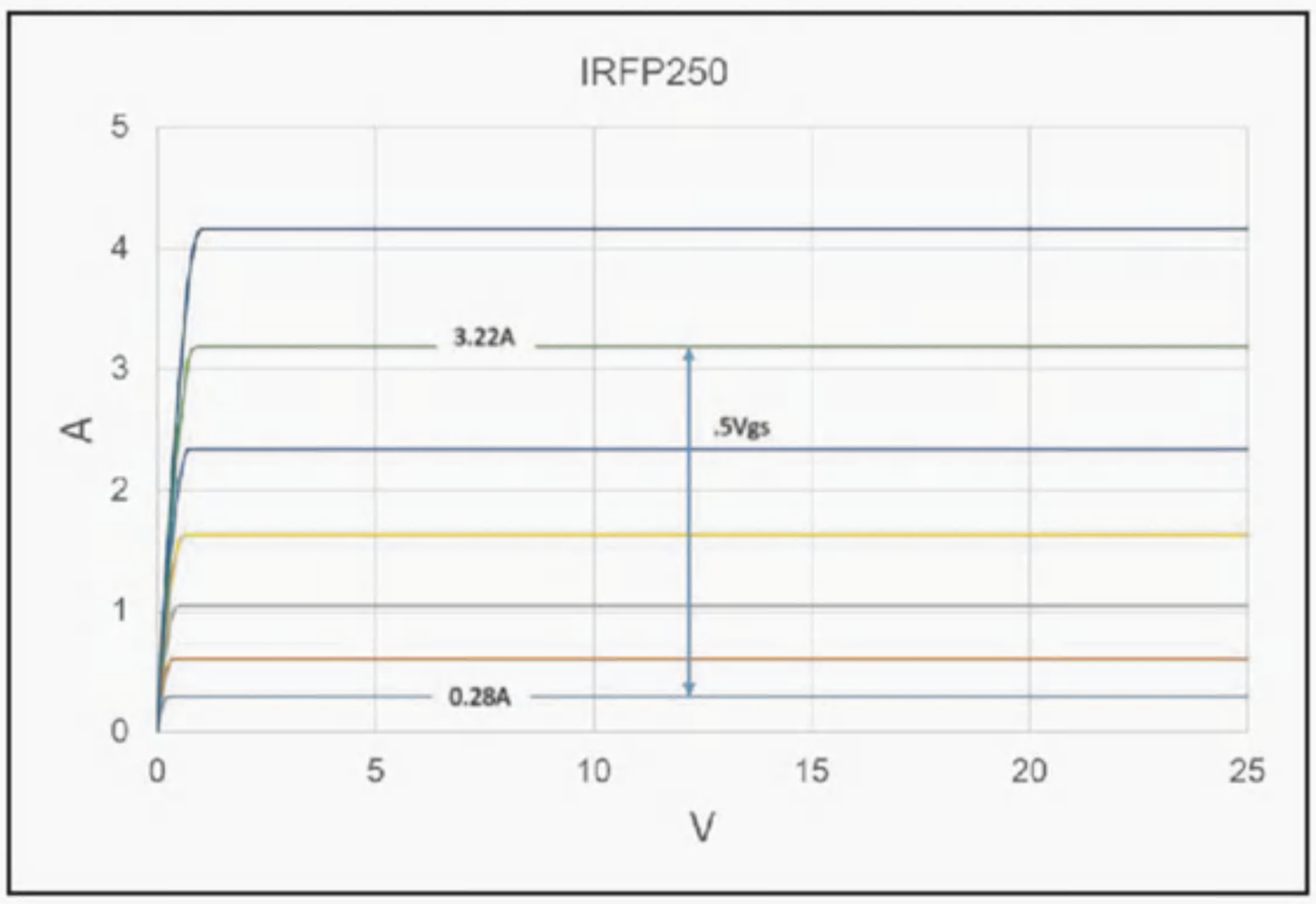


Figure 3: This graph shows a set of characteristic curves for an IRFP250 MOSFET.

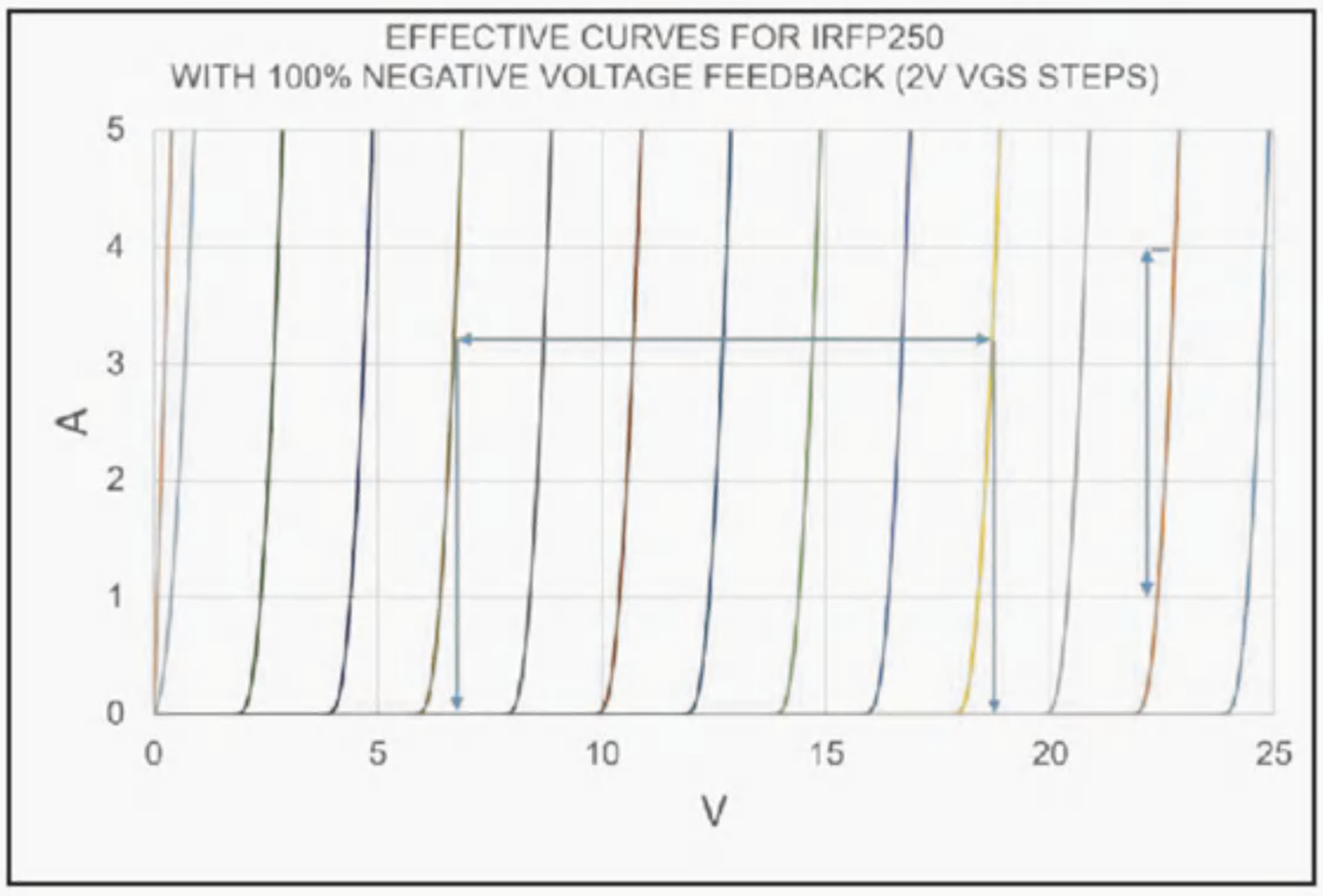


Figure 4: Here are the effective characteristics for an IRFP250 with 100% series negative feedback applied.

Well, the Source Follower can be thought of as a special case of voltage negative feedback where 100% of the voltage derived at the output is mixed back in series with the input. Like other forms of voltage negative feedback, it has a "triodey" effect on the characteristics of the gain device, making the IV curves more vertical. It will lower the gain, the distortion, and the output impedance.

Figure 4 shows the effective characteristics for an IRFP250 with 100% series negative feedback applied. Without any math at all we can see that we have low output resistance (vertical curves), low distortion (evenly spaced parallel curves), and little or no gain.

Since this indeed looks sort of "triodey" let's analyze Figure 4 in triode terms in order to understand the interrelationship between these two sets of IRFP250 curves. The output resistance for Figure 4 is the slope of the curve, which is the change in voltage divided by the change in current or:

$$rS = \frac{\Delta V}{\Delta I} = \frac{0.5}{3} = 0.17\Omega \qquad \text{EQ 1}$$

The amplification factor is the change in Drain voltage divided by the change in Gate voltage:

$$\mu = \frac{\Delta Vd}{\Delta Vg} = \frac{12}{12} = 1 \qquad \text{EQ 2}$$

OK, so next let's talk about transconductance, which in triode terms is mu over the output resistance:

$$g_m = \frac{\mu}{rS} = \frac{1}{0.17} = 5.88s \qquad \text{EQ 3}$$

And now we turn back to the curves without feedback applied. We don't normally think of MOSFETs as having mu, so we calculate the transconductance of Figure 3 as the change in output current over the change in input voltage:

$$g_m = \frac{\Delta I}{\Delta V} = \frac{2.94}{0.5} = 5.88s \qquad \text{EQ 4}$$

We notice that transconductance is unchanged between the two sets of curves. Transconductance is the same, but we've lowered the output resistance and the gain.

Now, we normally calculate the output impedance of a MOSFET Source Follower as the

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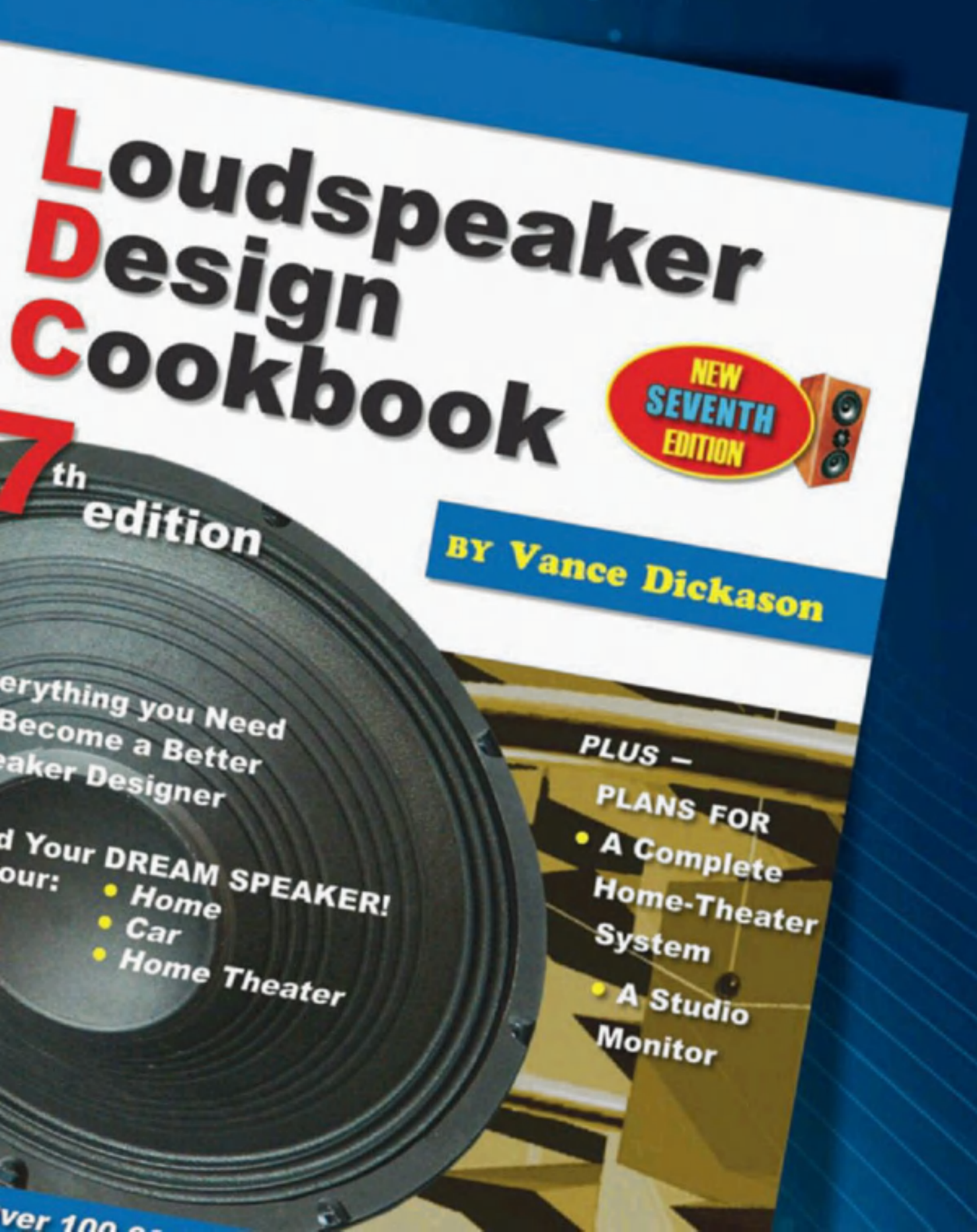
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inverse of the transconductance:

$$rS = \frac{1}{g_m} = \frac{1}{5.88} = 0.17\Omega \qquad \text{EQ 5}$$

And we see that's the same output resistance we get from our "triodey" effective curves.

So there we go, we have a different way of looking at what feedback really does and how it changes the gain device characteristics. Neat huh?

So, is The Dude saying this will sound just like a triode? No, not exactly. He probably wasn't being quite so literal, but it's likely to compare more closely to a Cathode Follower than a Common Cathode compares to a Common Drain. If I ever finish my Cathode Follower with 16 6AS7s, I'll let you know. Like The Dude said, you have to try it for yourself.

MoFo Measurements

The MoFo's input impedance is around 50 kΩ, and the output impedance is around 0.2 Ω, for a damping factor of 40. Noise (22 Hz to 22 kHz) was around 3 μV with the switching power supply I used.

Figure 5 shows the THD vs. Output Power in watts. This is with a 19 V power supply, biased at 1.7 A, and you can see we're making around 12 W. The Hammond 193T inductor will handle 2 A and you could consider taking the bias up a little for a little more power and a little less distortion, providing your power supply unit (PSU) and heatsink are okay with that. I left a little margin. The 12 V and 15 V curves (also at 1.7 A bias) are identical except, of course they clip earlier due to the lower rail. **Figure 6** shows the frequency response, a little less than 2 dB down at 200 kHz. **Figure 7** shows the THD vs. Frequency. Note the reduced (not rising) distortion at high frequency.

Figure 8 details the THD vs. Output Power in watts for the Big MoFo, the 24 V version biased at 2.5 A. You can see the higher bias nearly halves the distortion, and we're making it to 22 W or so.



Figure 5: The graph shows the THD vs. Output Power in watts.

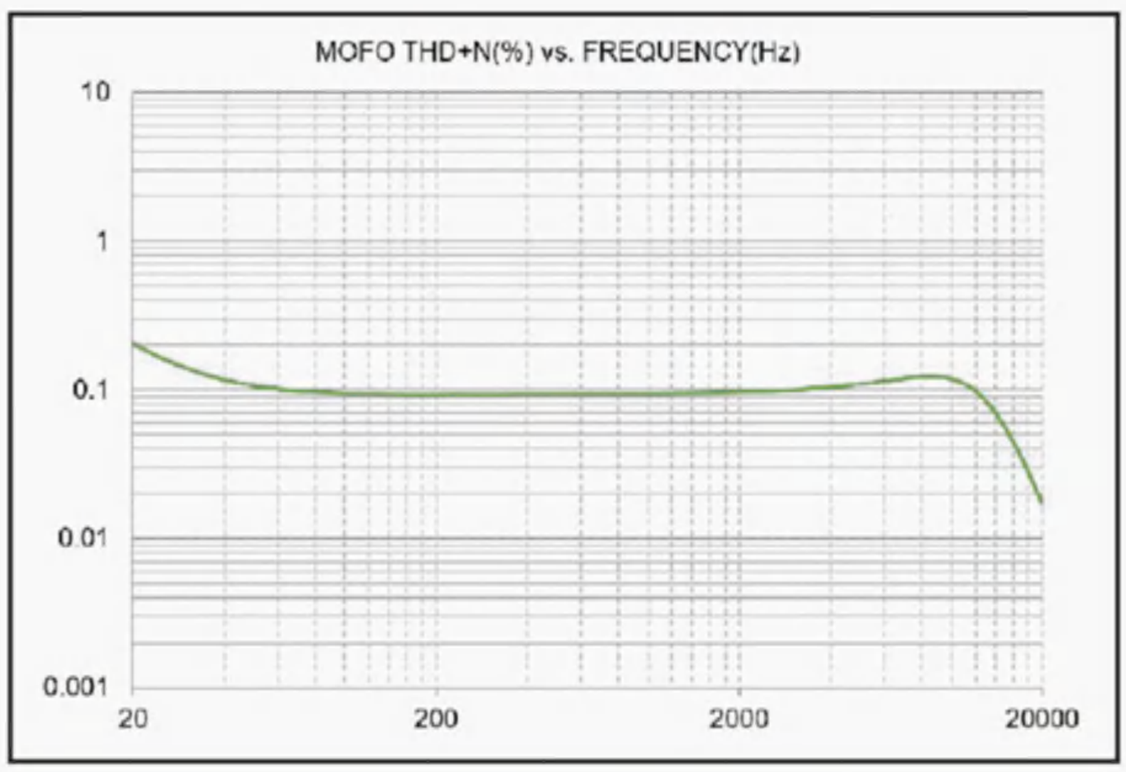


Figure 7: The THD vs. frequency response shows the reduced (not rising) distortion at high frequency.

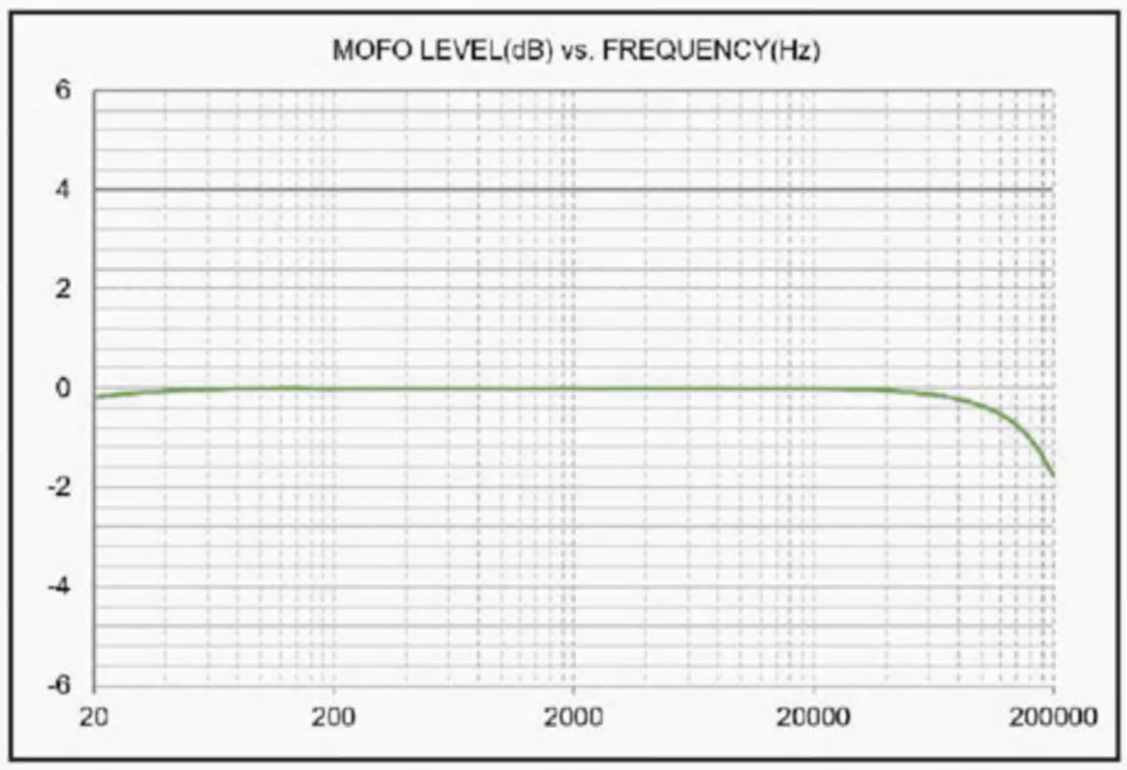


Figure 6: The frequency response is a little less than 2 dB down at 200 kHz

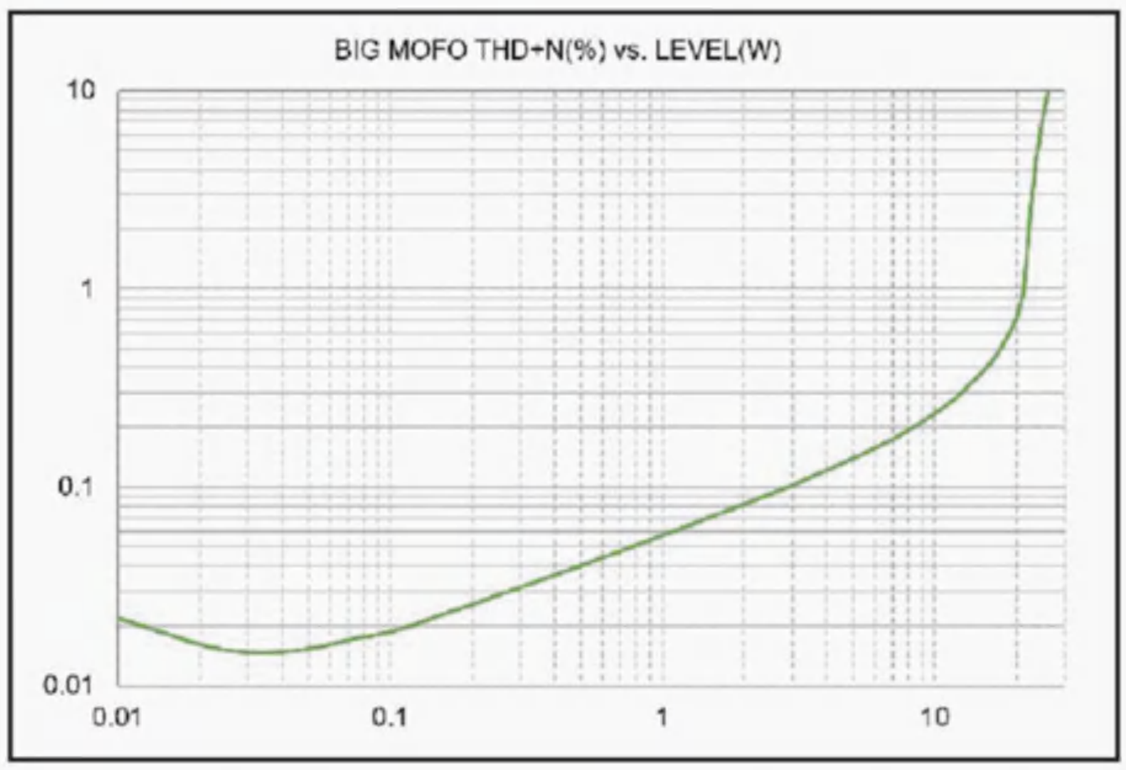


Figure 8: The THD vs. Output Power is provided in watts for the Big MoFo.

If this tempts you to build the big one, you'll need a bigger heatsink and the much larger Hammond 193 V inductor.

Figure 9 shows a 10 kHz square wave. And, finally, **Figure 10** is the THD spectrum at 1 W. You can see the second harmonic at about -60 dB and the audiophile-approved declining staircase of third, fourth, fifth, and sixth, vanishing into the noise floor.

Building MoFo

Table 1 details the parts list. The parts will cost approximately $50 per channel without the power supply and heatsink. These project articles usually draw a lot of questions about part substitutions and so on. Rest assured, I tried lots of different parts, and those on the list are what I recommend if you want your experience to be consistent with mine. On the other hand, I understand some of these parts might not be readily available in your area, or you really like to experiment, in which case I'll share a few thoughts on alternative parts.

You could substitute other MOSFETs, the old favorite IRFP240 for example, but the IRFP250 is just about optimal here, and they're dirt cheap. I know some of you might want to experiment with Q1. I tried lots of other parts. I tried fancy, expensive parts from Cree and they did not have lower distortion. I also tried obsolete parts from Semisouth and Fairchild and they had slightly better distortion performance, but they're unobtanium and of dubious merit here anyway.

If you try other MOSFETs and experience oscillation, you can increase the value of R2 until it stops. I found this to be the case with several of the low-capacitance Cree parts and took R2 up to 470 Ω or more. If you're trying to adjust the bias and things go all wibbly-wobbly, increase R2. Z1, a Zener diode, is there to protect the MOSFET's Gate. Disconnect power on the AC side of the power supply with a switch. If you just yank the DC jack out its socket, you're going to get a big turn-off thump, maybe big enough to damage your MOSFET. You might want to hardwire the DC side of the power supply rather than using jacks to minimize this possibility.

The Hammond 193T is just about ideal for channels up to 19 V. The maximum DC current is 2 A and I biased my 12 V, 15 V, and 19 V prototypes at 1.7 A to maximize the distortion performance. If you want to build a Big 24 V MoFo, you need the 193 V, which is larger, heavier, and more expensive, but it can handle the 2.5 A idle current.

If you can find a choke specially wound for audio, that would probably be better still, but it has to handle all the idle current with a little

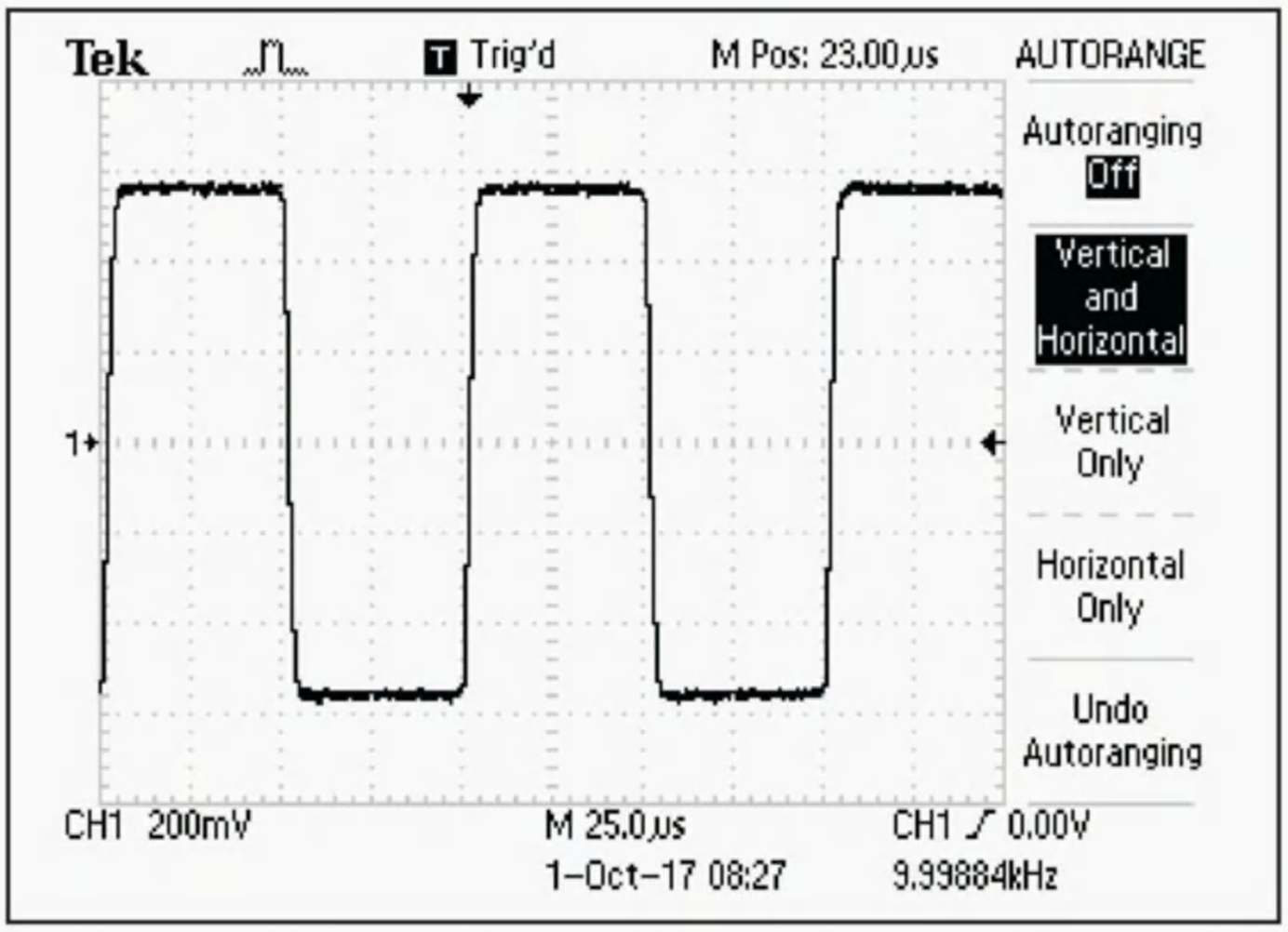


Figure 9: This graph shows a 10 kHz square wave.

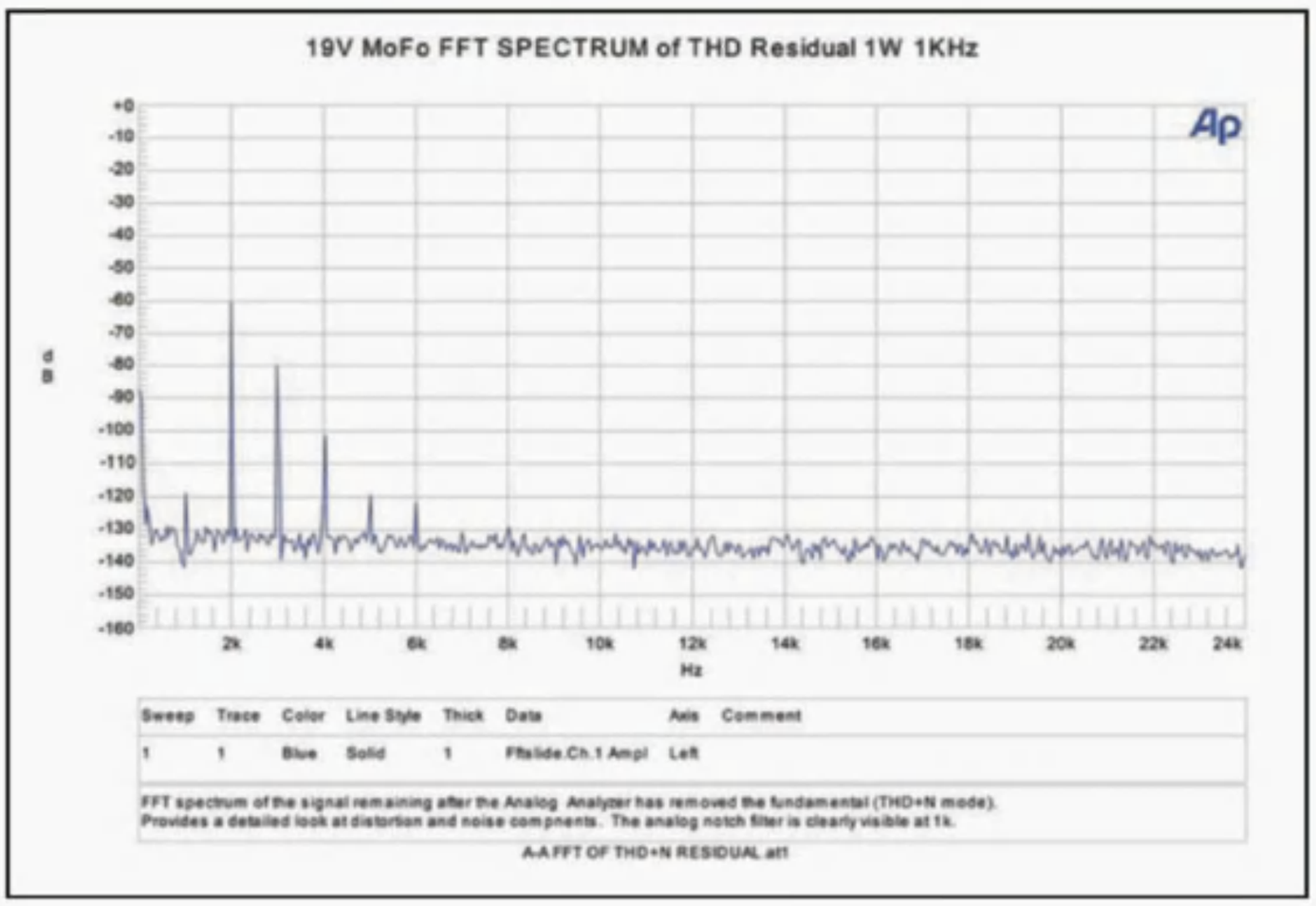


Figure 10: The THD spectrum is shown at 1 W.

Part ID	Value	Part Number	Quantity per Channel
R1	200 kΩ, 0.25 W		1
R2	221 kΩ, 0.25 W		1
R3	50 kΩ, 0.25 W		1
R4	15 kΩ, 0.25 W		1
R5	1 kΩ, 0.25 W		1
P1	20 kΩ trimmer	Mouser 652-3296W-1-203LF	1
C1	2.2 µF or greater film capacitor	Parts Express Part # 027-110	1
C2	6800 µF, 25, V	Mouser 647-UFW1E682MHD	1
Q1	IRFP250	Mouser 942-IRFP250NPBF	1
L1	50 mH Hammond 193T	Mouser 546-193T	1
Z1	20 V Zener diode	Mouser 512-BZX79C20	1
Power supply	19 V Switching	Mouser 709-GST90A19-P1M	1

Table 1: This is my recommended parts list for the MoFo project.

margin. You can try other chokes and inductors too. I tried smaller crossover-type (10 mH) air-core and iron-core types. If you try that approach you'll see significantly higher distortion at low frequency, which you might not want, on the other hand, it would make a fine tweeter amp. Ideally, you want your inductor to have low DC resistance so you don't drop a lot of voltage. The iron cores are attractive for this reason. The Hammonds were the best of the handful I tried.

Switching power supplies of the laptop variety come in fixed voltages like 12 V, 15 V, 19 V, and 24 V. You need one for each channel. I suggest finding some that are rated for 2 or more times the idle current. You can buy new ones, and there are also some good surplus bargains out there. Some are pretty junky. Be thrifty, but not too thrifty when selecting your power supply.

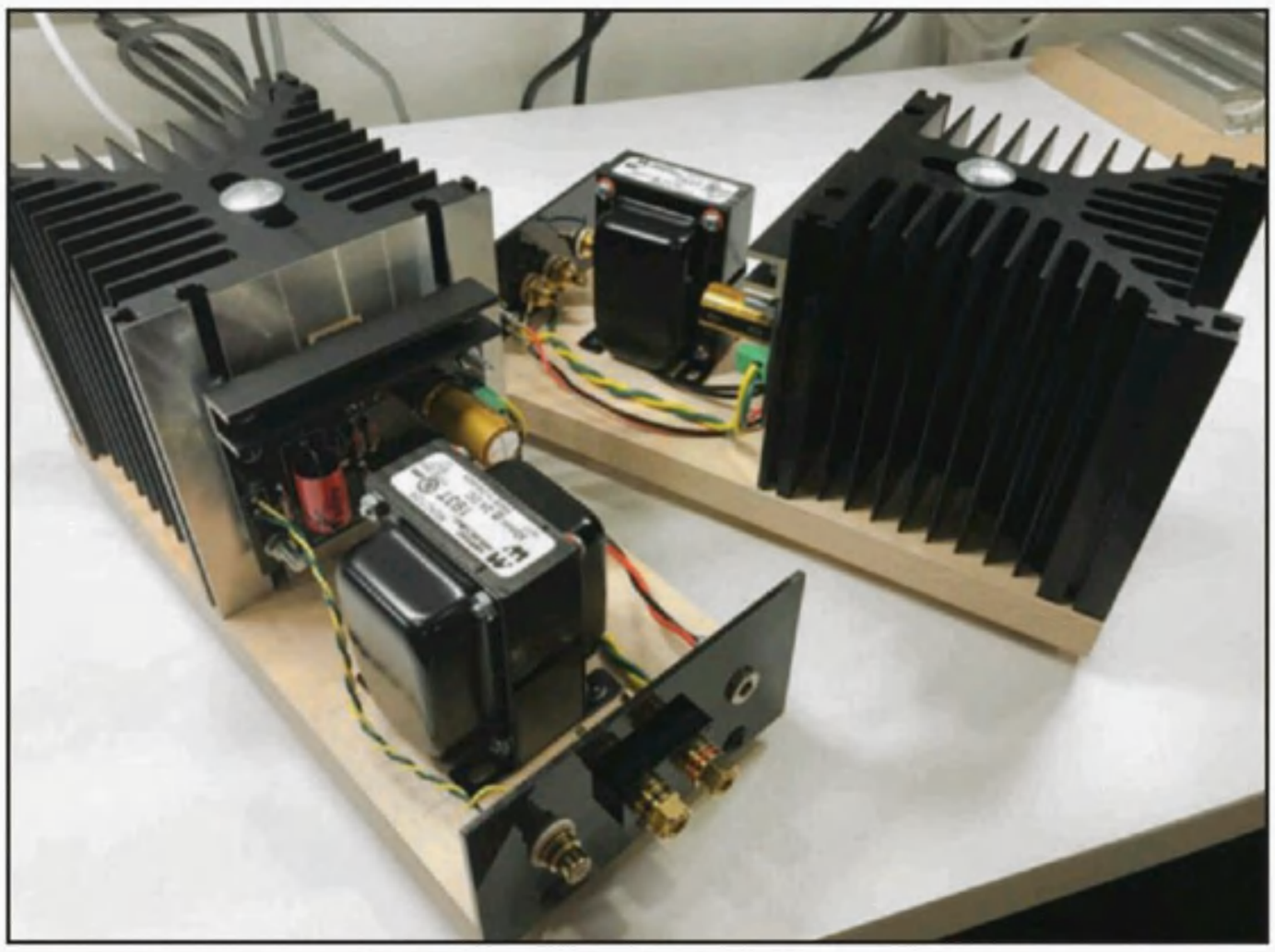


Photo 1: I built my MoFos as monoblock amps on plywood bases.

About the Author

Michael Rothacher is an average guy with a regular job, but his alter ego is a mad scientist. He caught the audio bug more than 20 years ago and has been building audio gear ever since. He proudly counts himself among the many students of Nelson Pass, the "Wizard of Sea Ranch" (to whom he happily attributes all the smart bits of his work) and enjoys exploring the boundaries of simple, high-performance circuits. He is a firm believer that many of life's problems can be solved with Ohm's Law, and encourages you to not worry so much about what other people think, try your crazy ideas, trust your ears, and never stop learning.

Project Files

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http://audioxpress.com/page/audioXpress-Supplementary-Material.html

The device dissipation will be pretty close to your power supply voltage times the idle current. So, for the 19 V, 1.7 A example, we need a heatsink that can dissipate 32 W with no more than a 25°C rise above ambient temperature. That's 0.78°C/W, which I consider to be the minimum figure. I will award bonus points if you use oversized heatsinks. A lot of beginner projects run into trouble because of inadequate heatsinking, and using big heatsinks just makes life easier.

You should build these as monoblocks. Monoblocks are cool. I built mine on plywood as seen in **Photo 1**. The single-sided PCB is for one channel of MoFo. These should be available for purchase somewhere online soon after this article goes out. These things solder-up very quickly, even if you're a beginner, you should be able build a pair in an afternoon once you have all the parts. Of course, I suggest you take your time and enjoy the building (see **Figure 11**).

Once everything is assembled, you're ready to adjust the bias. Before you attach the inductors, carefully measure and write-down the DCR of each. The Hammond 193T will be around 0.6 Ω and the 193 V will be around 1 Ω.

Next, adjust P1 for its minimum setting (fully counter-clockwise) and plug in your amplifier channel. You're going to set the bias at around 1.7 A, unless you're building the big one, which is biased at 2.5 A. Measure the voltage across the inductor terminals while you slowly adjust P1 upward. The current through the inductor is the voltage you measure divided by the resistance you wrote down earlier. So, the voltage reading you want is your target current times the DCR. For example, if your DCR is 0.6 Ω and you want 1.7 A through L1, you're looking for 1.7 × 0.6 = 1.02 V at the inductor terminals.

Alternatively, you can insert a 0.1 Ω 3 W resistor in series with the positive terminal of your PSU and measure the voltage there (e.g., 0.1 × 1.7 = 0.17 V).

Once you set the bias for both channels, you should observe the current for an hour or so, then re-adjust the bias with the channels fully warmed-up. If you're heatsinking is adequate, you should be able to place your hand on the heatsink without discomfort, if only for a few seconds.

More MoFo

Since MoFo is a power buffer it's actually a pretty versatile amp. For a comprehensive look at many interesting applications for power buffers, I highly recommend taking a look at First Watt's "Model F4 Operation and Service Manual," which can be found at www.firstwatt.com. It's one of Nelson Pass' most entertaining and informative manuals.

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COLUMNISTS

Vance Dickason has been working as a professional in the loudspeaker industry since 1974. He is the author of *Loudspeaker Design Cookbook*—now in its seventh edition and published in English, French, German, Dutch, Italian, Spanish, and Portuguese—and *The Loudspeaker Recipes*. Vance is the editor of *Voice Coil: The Periodical for the Loudspeaker Industry*, a monthly publication. Although he has been involved with publishing throughout his career, he still works as an engineering consultant for a number of loudspeaker manufacturers.

Dr. Richard Honeycutt fell in love with acoustics when his father brought home a copy of Leo Beranek's landmark text on the subject when Richard was in the ninth grade. Richard is a member of the North Carolina chapter of the Acoustical Society of America. Richard has his own business involving musical instruments and sound systems. He has been an active acoustics consultant since he received his PhD in electroacoustics from the Union Institute in 2004. Richard's work includes architectural acoustics, sound system design, and community noise analysis.

Mike Klasco is the president of Menlo Scientific, a consulting firm for the loudspeaker industry, located in Richmond, CA. He is a graduate from New York University, with post graduate work in signal processing, and he holds multiple patents licensed or assigned. For the past 35 years, Mike has worked on countless R&D projects for large and small companies. Mike specializes in materials and fabrication techniques to enhance audio performance.

Michael Steffes has worked for more than three decades in high-speed signal path IC development, spanning IC design roles for the BurrBrown high-speed amplifier products, and product launch/application manager roles through Texas Instruments/Intersil. He continues to focus on high-performance ADC interface solutions, DSL/PLC line driver design and optimization, and signal path design optimization tools. In a career spanning five different major IC vendors, he has participated in the introduction of 150 industry leading amplifier products (setting many of the now standard practices in high speed amplifier characterization curves), publishing more than 100 contributed articles and application notes from 1989 onward.



A few applications that I've tried include:

- You can bridge two MoFos. In fact, two Big MoFos gets you close to 100 W.
- You can bi-amp with passive crossover networks. I bi-amped my SAL08C08/Hawthorne Augie open baffles (see Dick Olsher's write-up). A L'Amp DeLux drives the full range and a MoFo drives the Augie. The two make a very nice combination.

So, you want to know how the MoFo sounds. Well, I'm biased, of course, but I think these are more fun than a barrel of monkeys and they sound really good. Paired with a capable preamp (and that's an important caveat), I think you'll be very, very surprised at their transparency, depth, and ability to cast a great big soundstage. You ought to build a pair of these and see for yourself. I plan to expand on this theme in the future, because there's even more fun to be had.

I'd like to thank The Dude for tapping into my subconscious and helping me out. I'm sure he's out there somewhere, taking it easy, building amps, and rolling strikes.

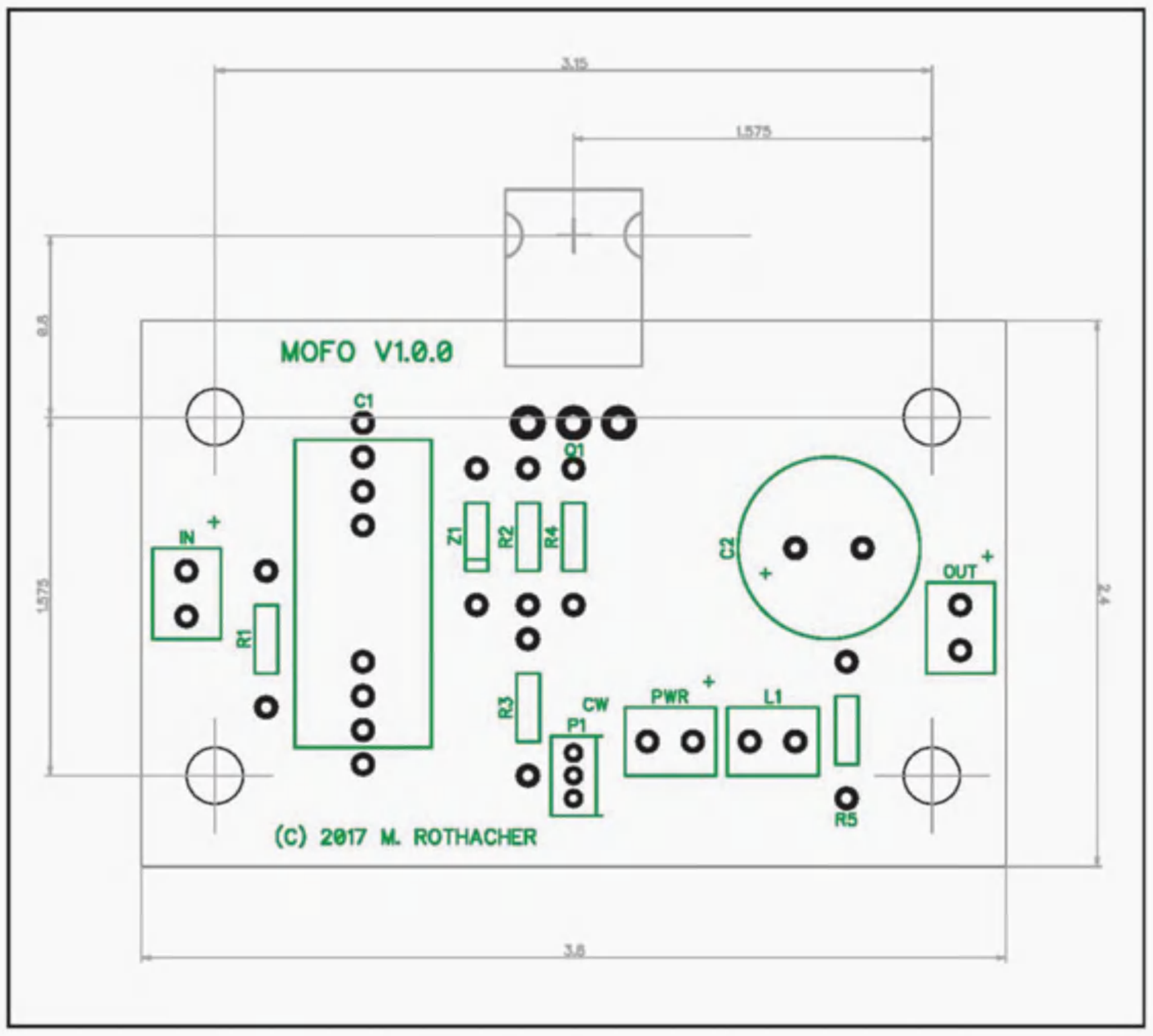


Figure 11: Here is the PCB layout for the MoFo project.

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By Robert B. Tomer

Vacuum Tubes—The Important 4/5ths of Tube Circuits

Over 80% of all tube equipment defects result, directly or indirectly, from tube failures. Why do tubes fail? What can be done to prevent them from failing before their time? How can you determine whether a tube is good or bad, or how well and how long it will work in a given circuit? Should tubes be replaced periodically, whether they've failed or not . . . or should they be tested every so often, and replaced if indications show them to be below par? This book supplies the answers to these profound questions…plus many, many more.

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In Memoriam of Gerhard Haas

(1/17/1952 - 1/28/2020)



audioXpress received notice of the passing of our long-time author and contributor, Gerhard Haas, from South Germany, at the age of 68. We would like to express our sincere condolences to his wife Sabine, daughter Christine, son Maximilian, and his grandchildren.

I met Gerhard Haas for the first time during an annual Elektor expert meeting somewhere in Germany, where he arrived together with his long-time friend Michael Kaim, the founder and owner of BTB Elektronik, Europe's largest tube supplier. It didn't take me long to understand that I had just been introduced to two of the world's most experienced and enthusiastic tube experts. At the time, I had just started to work for *audioXpress*, and we quickly started exchanging ideas on how to expand the editorial coverage from high-end to other tube audio applications, and explore the fascinating world of lesser known tubes.

Shortly afterward, Gerhard submitted an article about the Wah-wah's effect pedal (which he originally signed as "Paul Hendrix"), quickly followed by another one about the KT 88 and KT 120 power tetrodes, and another about a Class-A headphone amplifier. And we established a plan to translate more than 10 different articles from German to English, most of which were previously published by Elektor in its Roehren Sonderheft editions. Gerhard had written most of the articles for these Elektor "Tube Special Edition" magazines, including bylined articles using an alias, but the articles had never before been available in English.

Gerhard's talent and extraordinary work was the pure result of dedication and passion—he described himself as an amateur, and always insisted that his projects should be available as low-cost DIY kits so others like him could also learn. But he certainly had a natural instinct for audio electronics and a contagious enthusiasm for high-quality audio experiences, always considering the complete chain, always powered by tubes. "Whether guitar amplifier sound or hi-fi, a second-harmonic-focused amplifier sounds better," he wrote.

I know that much of the richness and detail of Gerhard's works never fully translated from his original text in German to the published version in US English published by *audioXpress*. The German language represents a significant challenge for many publishers and always requires carefully translated work.

Sometimes we would be dumbfounded to read paragraphs of text with a less than obvious inherent humor. Describing the Russian 6C33 tube, Gerhard wrote: "Such a high degree of resistance to vibration and shock is perfectly appropriate for the use in fighter jets, tanks, or warships. Therefore, this tube could be used for beating people to death without damaging it. These characteristics are less important for hi-fi purposes."

But Gerhard Haas' work expands way beyond. His published work includes many more articles for *Elektor* and other magazines, and two books published, the first being his popular *High-End mit Röhren (High-End with Tubes),* first published in 1995, updated in 2005, and updated again in 2017, and the more recent *Messtechnik fuer Niederfrequenzverstaerker (Measuring Technics for AF Amplifiers),* published in 2018.

Discussing his book *High-End mit Roehren*, Gerhard always hoped that his circuits, which make up about half of the book, and the theory part of the book would help others like him: "This is also the most valuable part of the book for me, because it teaches amateurs like me a great deal about the knowledge of electronic circuits."

In Germany, the Experience Electronics website (his former company) still promotes his Classic Series amplifiers. Some of his tube amp and preamp projects were published in the first years of *audioXpress* and before that in the pages of *Glass Audio*, where his amplifiers were also frequently advertised, stating:

"manufactured in Germany by Gerhard Haas, technical designer and builder of Sennheiser's Orpheus tube headphone amp." And that's a story that still needs to be written, but Gerhard was always too modest to share the details.

—J. Martins
(Editor-in-Chief)

Gerhard Haas worked on many of his electronic projects in collaboration with Norman Weber (Weber Electronic), which shared more details about Gerhard's life and work, and sent us an important message from his son, Max Haas, which we share here. His friend Michael Kaim also contributed to this tribute.



One of the Classic Series Experience tube amps, which is also featured on this edition's cover as a tribute to Gerhard Haas.

We've known each other for longer than 30 years. We worked together on a lot of my own developments. And for some months already, we worked together on Gerhard's latest amplifier series. I started to work for Gerhard about 30 years ago on the Sennheiser ORPHEUS "reference" project. This was a very interesting experience, and after the project was completed, I continued working for him as a freelancer, helping to build his Experience amplifiers (before the company was sold) and also developing and assembling modules for mixers, etc.

A few years ago he had the idea to build a "final edition" of tube amplifiers, implementing the experience of the recent decades developing amps, transformers, and so forth, all in an exquisite chassis.

Together with his son Max we'll try to continue his lifework. The plan is to release several types of mono amplifiers (currently populated with EL34, KT88, and KT120).

—Norman Weber

My father was self-taught, but he took it to another level. He became interested in electronics and amplifiers while very young and, as a 15 or 16 year old, he could already easily repair radios, TVs, amplifiers, and more. He acquired a deep knowledge of math, physics, chemistry, and electrical engineering over the years and decades. He was also always eager to expand his knowledge. He summarized his extraordinary skills about high-end amplifiers in his clearly understandable book *High-End with Tubes*.

He set up his own business in 1985. Tubes always received his special attention due to the "warm," pure and unadulterated sound that could be achieved. The Orpheus HEV90 vacuum tube headphone amplifier, which he developed in 1990-1991 was definitely one of his "official" masterpieces, and his significant participation in the project was instrumental, even if he never claimed credit for it. I am not surprised, however, since my father was a very humble person and reluctant to show off his achievements.

For me it is extraordinary that this development happened in his early years of work as a freelancer, and that at that time he was already able to create an amplifier, which in combination with the HE90 (Sennheiser electrostatic headphone), set new standards for decades—and probably also beyond.

In addition to amplifiers, he developed custom-made products on customer request. In addition to his numerous published technical articles and books he also often developed projects for recording studios. His knowledge and willingness to optimize technology to the limit of what is physically possible never ended. Developing was his real passion.

After he sold his company Experience Electronics in 2009, he retired for a few years, to start the development of a brand-new platform. He called the new amplifier series his masterpiece.

We as a family, together with Norman Weber, intend to offer this masterpiece under the name of LINOS to audiophile customers looking for authenticity and clarity. Norman is an essential resource, for he is fully familiar with my father's amplifiers and was involved in this project from the very beginning.

For me, it is very important that my father was an honest, upright person. He was always fully transparent with regard to the quality of the amplifiers, and always specified reproducible, measurable parameters that reflect the quality and enable comparability. His standard was always to offer a fair product, so that users can enjoy one of the most beautiful things in the world, unaffected by technology—music!

—Maximilian Haas

Gerhard Haas was not just a customer. I called him a friend for many, many years. We often met at his house or in our home, and at the BTB company (only 100 miles away from him). He was always present to talk and discuss audio business, which also was a hobby. I have to say, thank you Gerhard and I will help Norman and Max to carry on with your Masterpieces!

We will keep you always in our minds.

—Your Friend Michael Kaim

Repurposing an Old Preamp for Line-Level Duty

By
Bill Reeve
(United States)



Photo 1: The Shure Brothers Model 65 stereo conversion preamplifier. Note the two, shielded 12AX7 tubes, the old capacitor can (which I disconnected but retained for aesthetics) and the small power transformer with B+ and heater output windings. The switch on the left selected various amplifier paths (tuner, mic, tape). It would be replaced by a volume control.

Bill Reeve builds a tube preamplifier to position at the input to his home power amplifier, eliminating the need to adjust his 1972 McIntosh MX10 tuner and 240 power amplifier's volume high when he wants to fill his home with music.

For many years, I have wanted to add a low-gain, (2× to 5×) line-level preamp at the input to our home power amplifier. The relatively low output voltage amplitudes of the switched audio inputs (a turntable and Bluetooth receiver) required adjusting the 1972 McIntosh MX10 tuner and the 240 power amplifier's volume high if I want to fill the house with music.

Placing a solid-state preamp between the switched line-level audio inputs and the old tuner and power amplifier would be acoustically clean, but building a tube preamp always seemed more appealing. The effort involved in constructing all the mechanicals for a stereo tube preamp, combined with the fact that this was not a pressing problem, kept the project off my priority list for years.

Eventually, I realized that I had a flea market find that could supply the mechanicals. It was a small, 1960s Shure Brothers stereo preamplifier (Model 65) that had been in the closet for decades (see **Photo 1**). Mechanically, it was beautiful, but the channel gain was extremely high. However, the little preamp had a chromed case with shielded tubes, audio connectors as well as a small B+ and heater transformer built in. It would be a perfect mechanical platform for the preamp I wanted.

The new preamp design would be relatively simple because it was receiving a low-impedance signal and driving a high-impedance load. The old Shure preamp used two 12AX7 dual-triode tubes (one for each stereo channel). The schematic for the original Model 65 preamp and datasheets for the 12AX7 tubes are easy to find online. In Shure's original preamp, each audio channel was a series pair of common-cathode amplifiers (because the design needed lots of gain). I wanted very little gain, so an obvious architecture for the new preamp was a common-cathode amplifier (for gain) followed by a cathode-follower stage (to lower the output impedance).



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Amplifiers



Vacuum Tubes



Capacitors



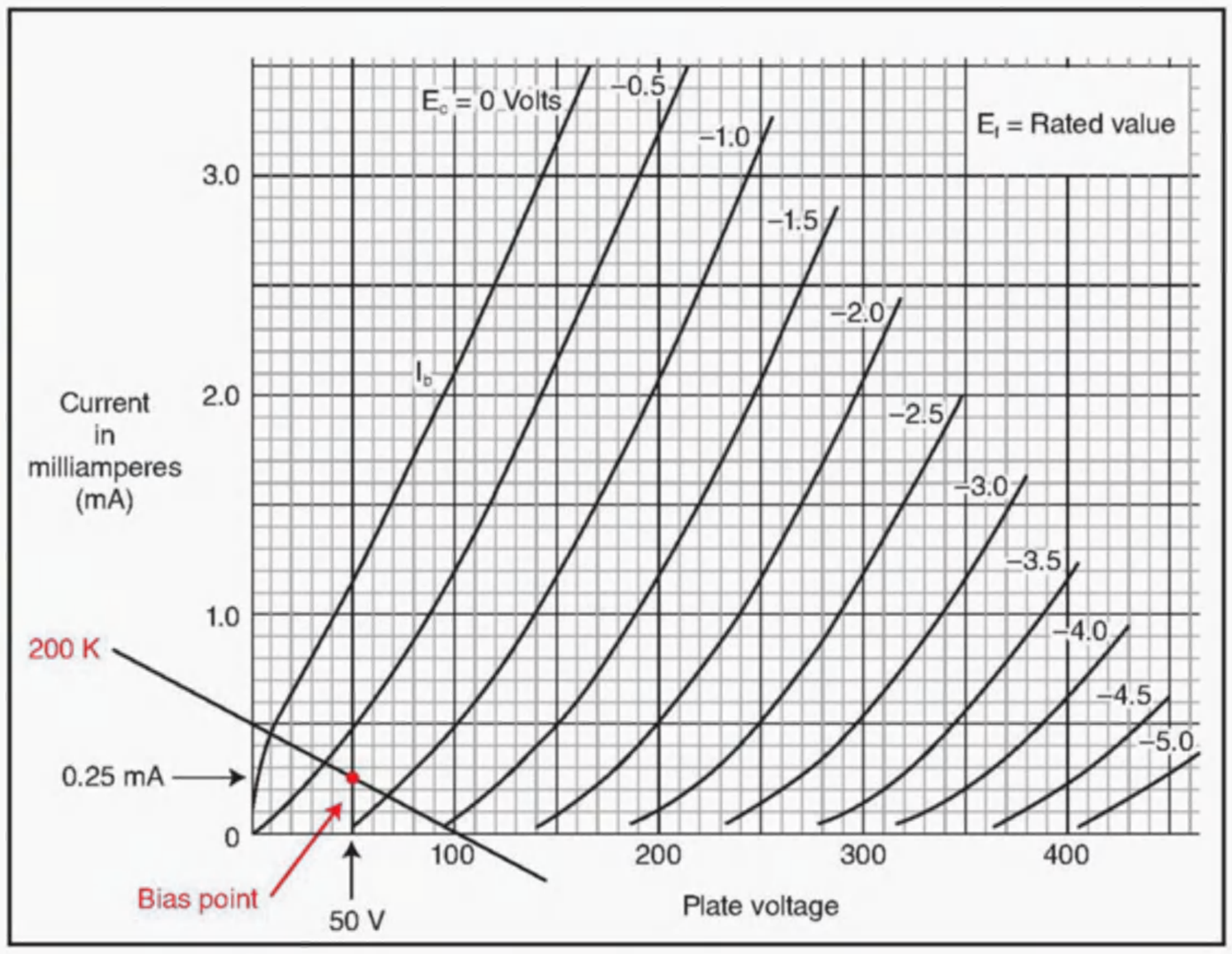


Figure 1: I located the common-cathode amplifier nominal operating point on the 12AX7 datasheet's "average plate characteristics" graph.

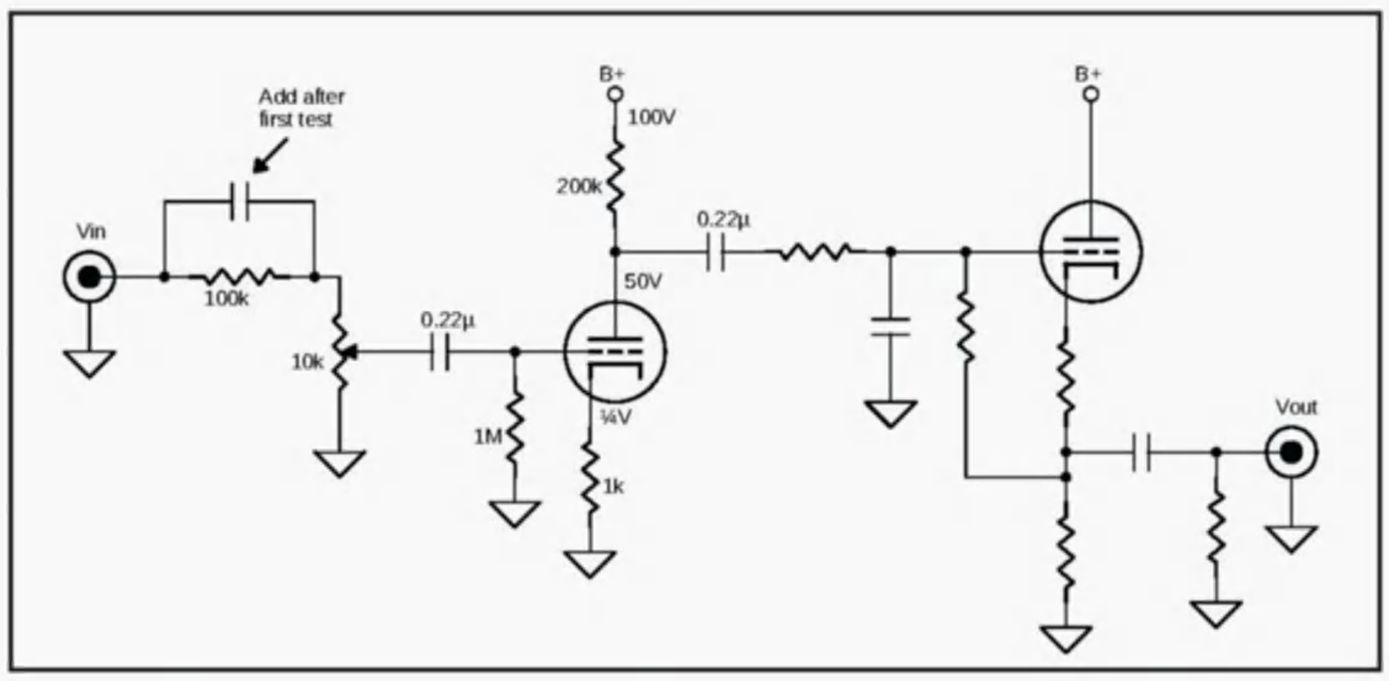


Figure 2: This is my initial design concept prior to building the first preamp stage.

The Creative Process

Because common-cathode amplifiers produce much more gain than this application needed, ironically, it was necessary to reduce the level of the input signal before sending it into the preamp. This initial gain reduction would be accomplished by a potentiometer at the input. This volume knob could occupy the spot originally filled by the Shure amplifier's channel-select switch.

The 1967 Shure schematic shows that B+ is an unregulated 100 V. I wanted a nominal plate voltage of half of B+ to maximize headroom. Since the power transformer is small, I didn't want to exceed 0.75 mA per channel (the original circuit used 0.5 mA per channel). I would need to save most of this 0.75 mA for the second (output) stage, so I allocated 0.25 mA to the first amplifier.

After locating an operating point of 0.25 mA and 50 V on the "Average Plate Characteristics" graph in the tube datasheet (see **Figure 1**), it was obvious that a 200 kΩ load line connected nicely with the maximum plate voltage (B+) of 100 V.

Since this first stage will be feeding the high-impedance grid of the cathode-follower second stage, the external load on the first-stage amplifier will be primarily its 200 kΩ plate resistor. A 12AX7 datasheet gives the following typical tube characteristics:

mu (tube gain) = 100
ra (internal plate resistance) = 80 kΩ

So, the low frequency gain should be approximately:

$Av = mu\ (R_{load})/(R_{load} + ra)$

Substituting 80 kΩ for ra, 100 for mu, and

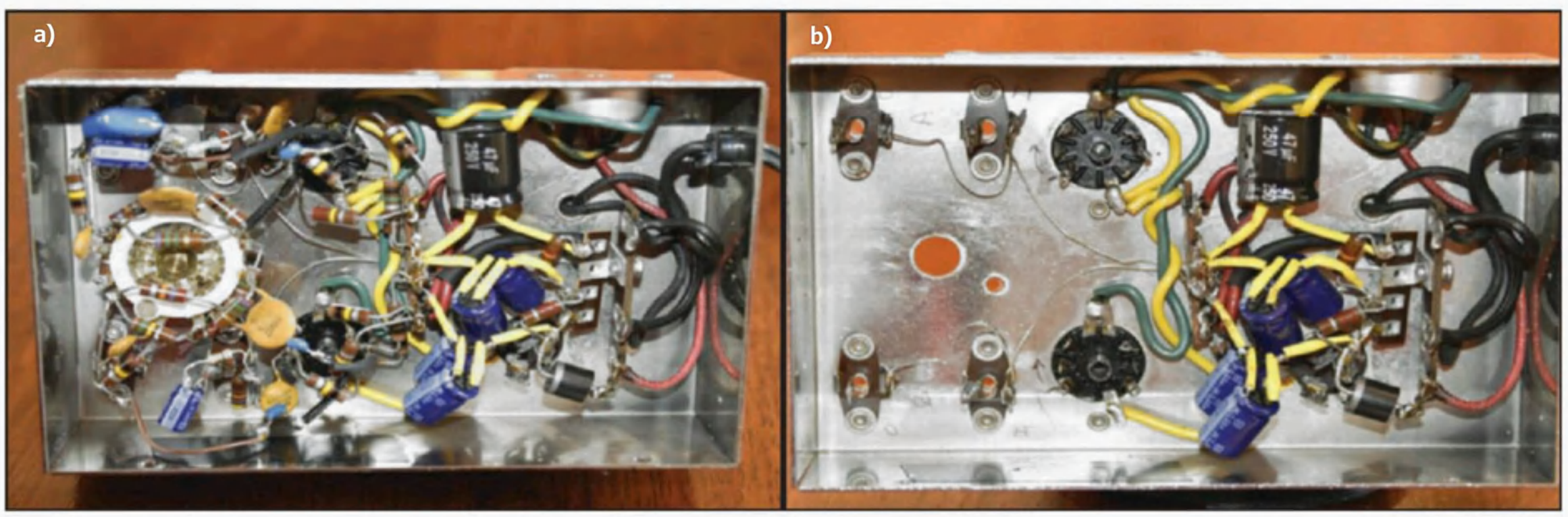


Photo 2: The original enclosure is shown (a) and with the original signal-path circuit removed (b). The power section is on the right side of the enclosure.

200 kΩ for R_{load} yields an approximate low-frequency gain of 70. Clearly, the input gain-reduction potentiometer was necessary, and in fact, we need to add a series resistor on top of the potentiometer to allow it to nominally work near the middle of its range at our desired gain of ×2. Because this resistor calculates out to be 100 kΩ for a gain of ×2 (33 kΩ for a gain of ×5) at the potentiometer midpoint, we will need to bypass it with a small capacitor. Stray capacitance, will create a high-frequency pole that will roll off the amplifier's high-frequency response. I'd like to add a zero with this small capacitor at that pole frequency to extend the first-stage frequency response. My plan was to measure the amplifier's frequency response without this capacitor, determine the high-frequency pole, then calculate and add the capacitor.

To finish the first-stage design, I somewhat arbitrarily selected a 1 MΩ grid leak resistor and a 0.22 µF input-coupling capacitor. This combination should yield a high-pass filter (low-frequency zero) at about 4 Hz. **Figure 2** shows what the design concept looked like at this point.

It was now time to build and test one of the common-cathode stages. But first, I needed to remove the old circuit, leaving the B+ and heater portions. **Photo 2** shows the enclosure with the old circuit removed but the power section retained. The right side is the power section. Many years ago I had replaced the old wax and paper capacitors.

Testing the A-side first stage revealed clean performance. The plate and cathode voltages were as predicted. With the 10 kΩ input potentiometer set at its midpoint the gain was flat at a factor of 2 (6 dB). There was one low-frequency zero (as we expected), there was one pole at about 4 kHz and no other poles or zeros out to 100 kHz.

Now we could determine the capacitor value to place a zero at 4 kHz. The calculated value was 400 pF. I installed a 470 pF capacitor, and the

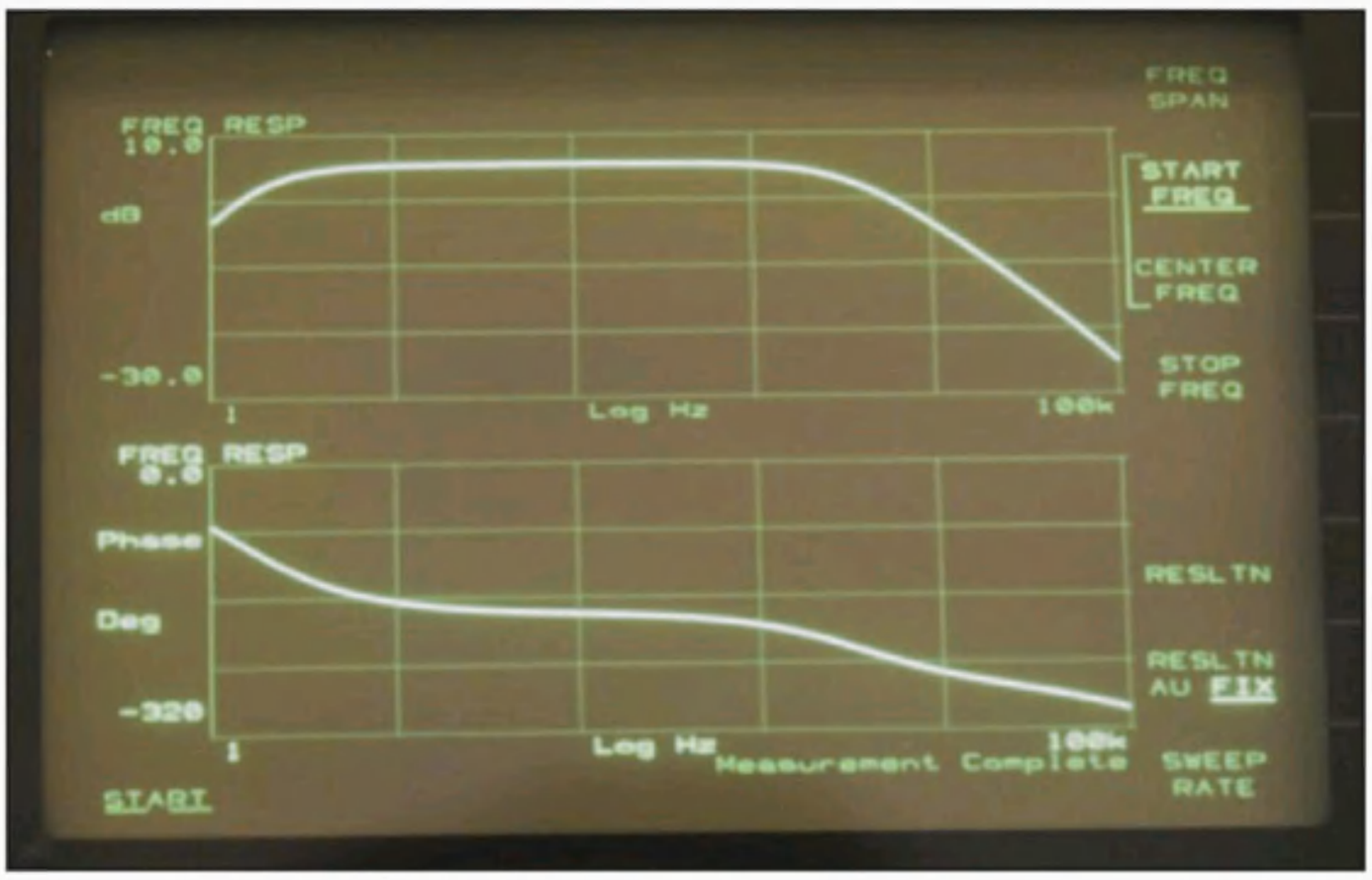


Figure 3: I installed a 470 pF capacitor, and this is the resulting frequency response.

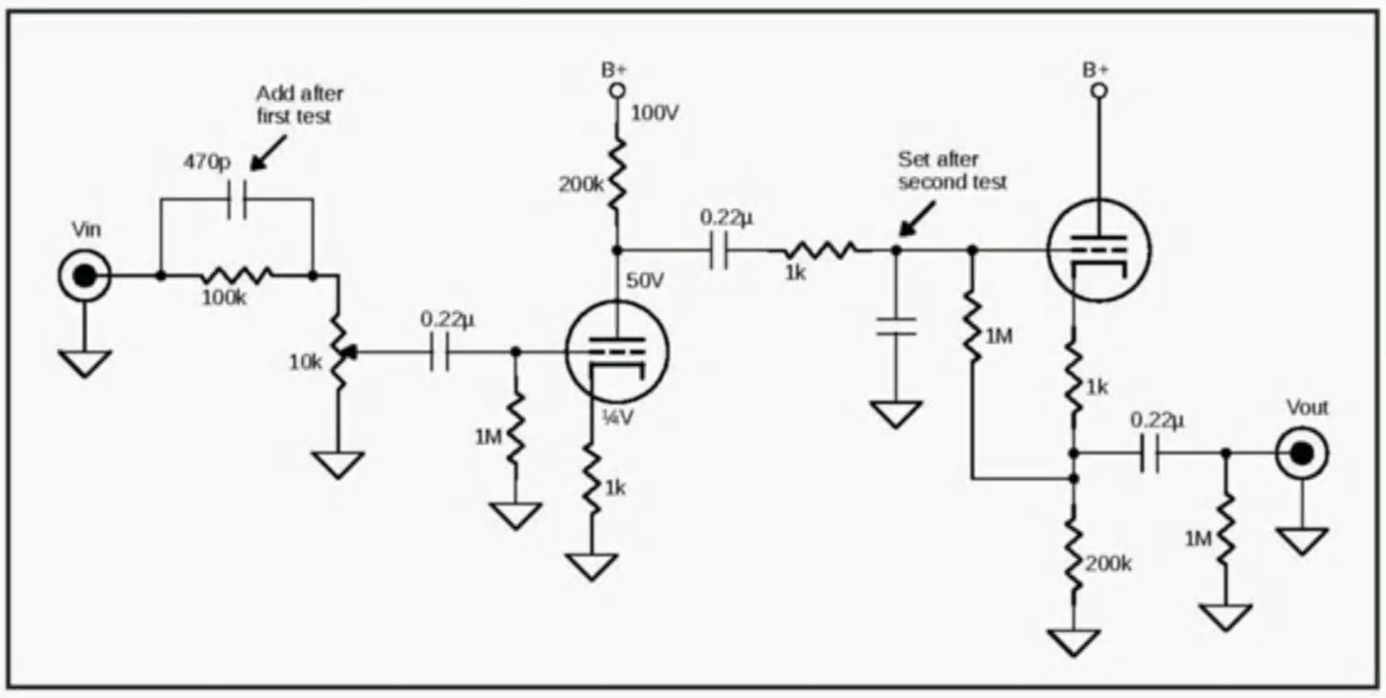


Figure 4: Here is my design concept prior to building the second preamp stage.

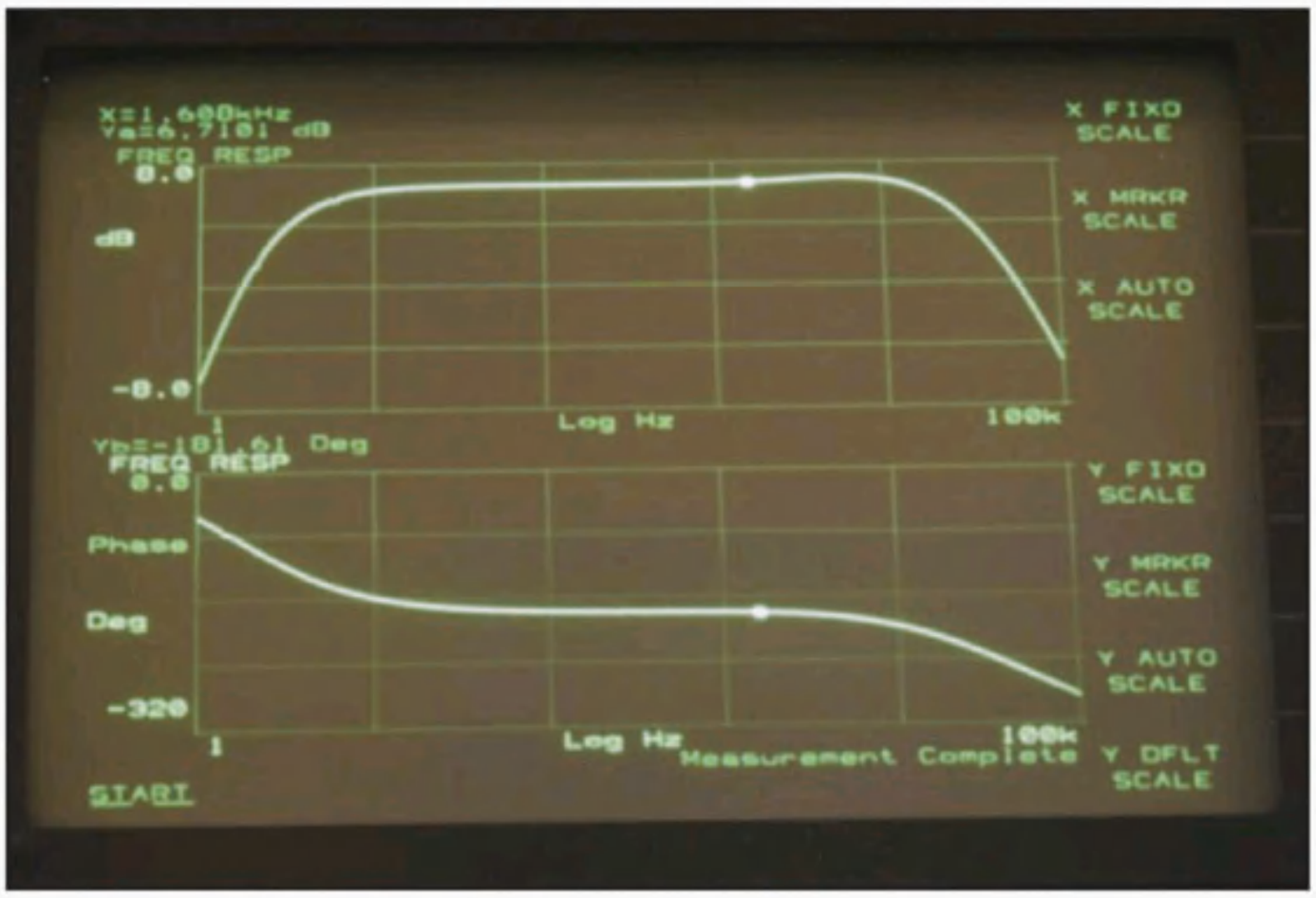


Figure 5: This is the complete end-to-end transfer function.

About the Author

Bill Reeve has a life-long interest in electrical and mechanical design. He has Masters of Science degrees in Electrical Engineering, Mechanical Engineering and Geological Engineering. He worked first as a consulting geologist, then spent 28 years at Lockheed Martin as a program director managing the development of NASA satellites and space instruments, in addition to serving as an Engineering Director at the DeWalt Industrial Tool company. Currently, Bill is a Technical Program Manager at Google.

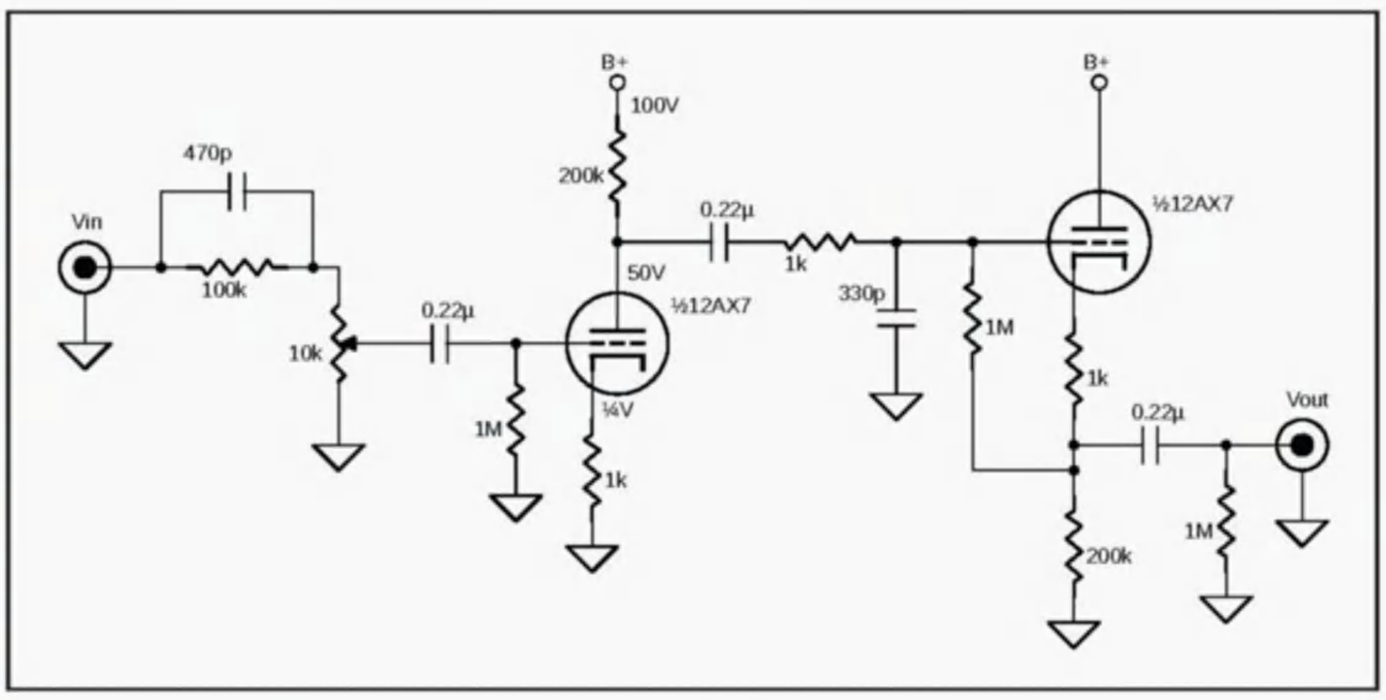


Figure 6: After all the configuring, this is the final design schematic.



Photo 3: This is the completed preamp. Note the dual volume adjustment potentiometer that replaced the selection switch.

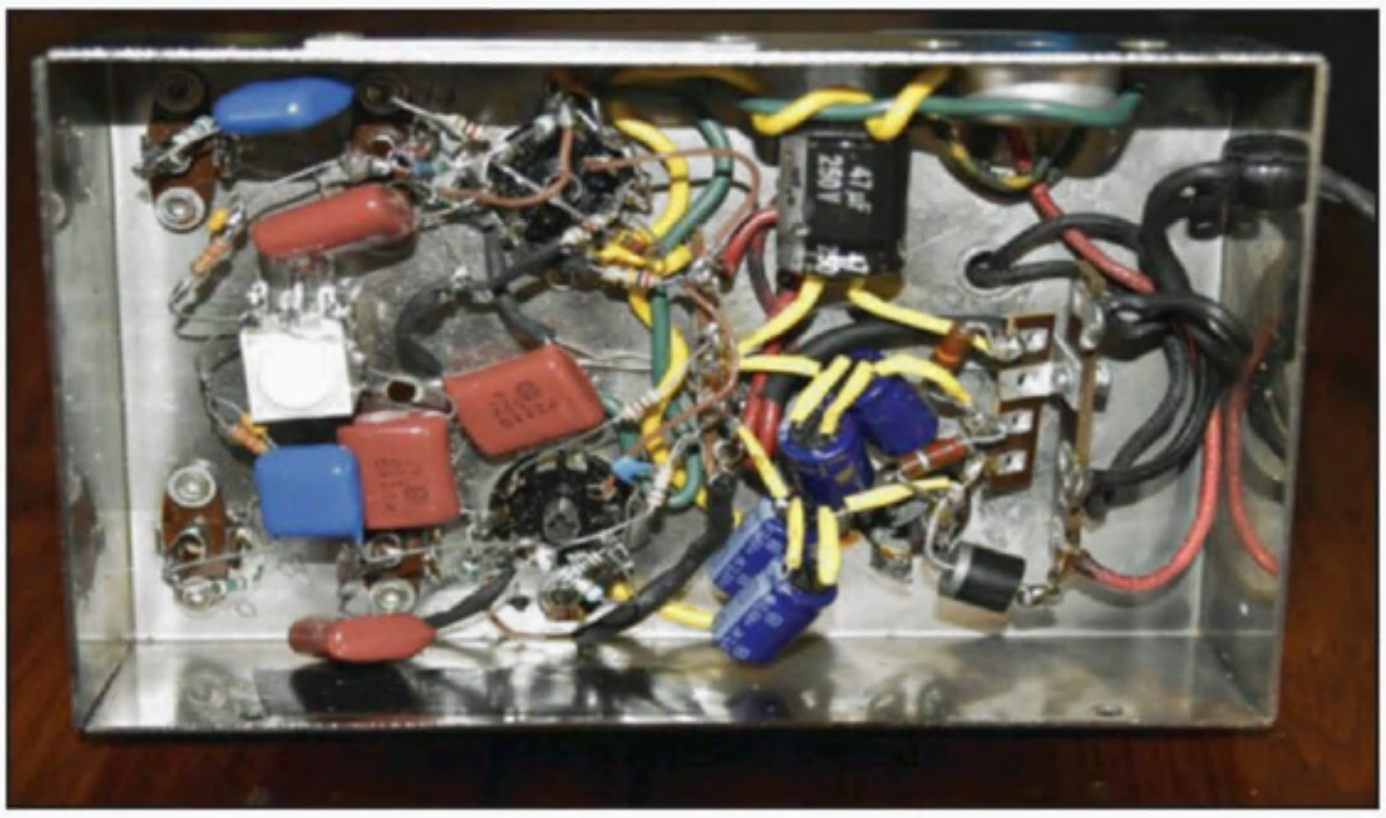


Photo 4: The inside of the amplifier with a dual (but not ganged) potentiometer replacing the original switch.

resulting frequency response is shown in **Figure 3**. These voltage transfer functions were measured using a Hewlett Packard 3562A Dynamic Signal Analyzer, but they can be plotted manually by recording the output/input voltage ratio at a number of frequencies. You can see the gain rising to the low-frequency zero, flattening off nicely through the audio range, and then rolling off with the added high-frequency pole. The first stage looked good.

Second Stage: Cathode-Follower Amplifier

The second stage is somewhat of an upside down version of the first stage. We want a 0.5 mA plate current. Using the calculation of 100 V divided by 0.5 mA yielded a cathode resistance of about 200 kΩ. So now we have an almost complete design. **Figure 4** shows the revised schematics at this point.

The only component remaining to be sized is the low-pass filter capacitor at the second grid. The cathode-follower will add some high-frequency dynamics to the transfer function, and the low-pass filter (aka a dominant pole) created by the yet-to-be-determined capacitor is intended to roll off the transfer function at about 20 kHz. This should eliminate any high-frequency shenanigans. An added 330 pF capacitor resulted in a very nice looking transfer function (see **Figure 5**).

Figure 6 shows the final schematic. **Photo 3** shows the completed amplifier. **Photo 4** shows the inside of the amplifier with a dual (but not ganged) potentiometer replacing the original switch.

The Result

This little preamplifier project looks great and I kept the original Shure Brothers nameplate. Finally, and most importantly, the little amplifier sounds nice.

Bill of Materials

Capacitors (250 V)	Quantity (Two Channels)
330 pF	2
460 pF	2
0.22 µF	6
Resistors (0.25 W)	
1 kΩ	6
10 kΩ potentiometer (dual)	1
100 kΩ	2
200 kΩ	4
1M	6

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GLASS AUDIO

Mini Blocks—Small Vacuum Tube Power Amplifiers

By
Bruce Heran
(United States)



Photo 1: In this latest iteration of my EL84 design, I built two amplifiers as separate mono blocks as they fit my present system needs better.

An Evolutionary Project Spanning 11 Years

Back in 2008 Bruce Heran designed and built a small tube-based power amplifier for his office. It worked far better than he expected and he used it for seven years. Although it still works just fine and has the original tubes, over the years, he has updated it a couple of times. This latest incarnation of the amplifier is actually a pair of mini blocks as they better suit his current needs. The mini blocks are quite similar in design to the original amplifier, but he has made a number of modifications that enhance the performance.

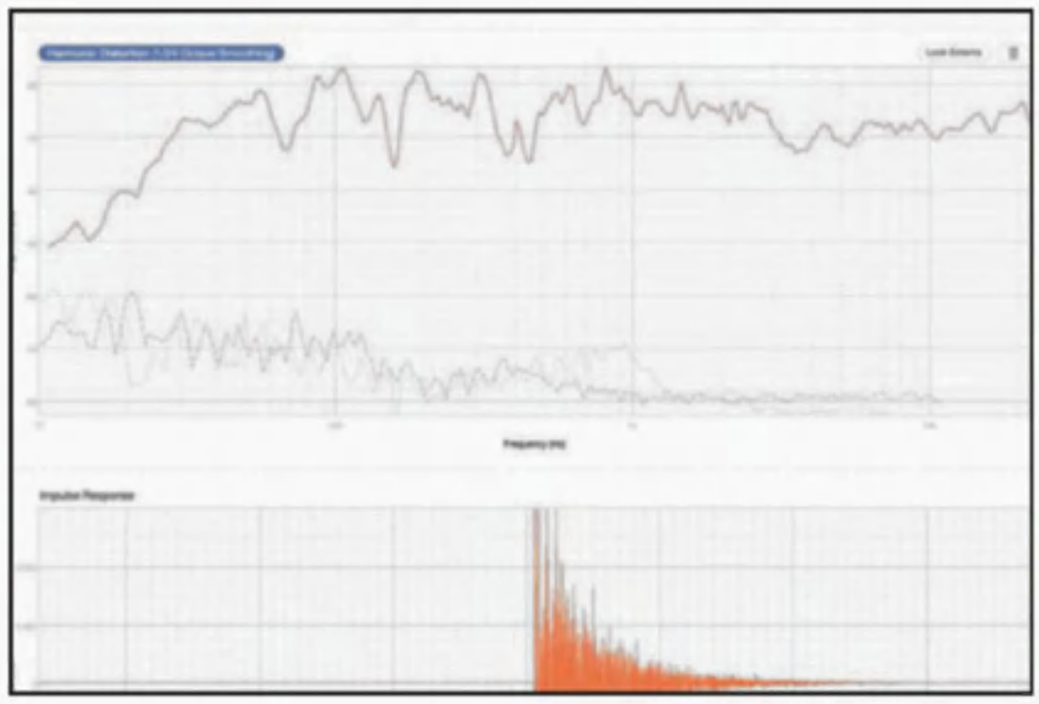
For this most recent evolution of my EL84 design, I built two amplifiers as separate mono blocks as they fit my present system needs better (see **Photo 1**). The power output is still the same at just over 5 W RMS but the sound is noticeably nicer. The new version has some features that are a bit non-standard but when considered in the context that I wanted the best sound I could get without breaking the bank, they make sense. Purists will holler at using an AC to DC Switch Mode Power Supply (Meanwell PM05-12) to power the heater of the driver/preamplifier tube, but it simplifies the build and makes the amplifier a few decibels quieter.

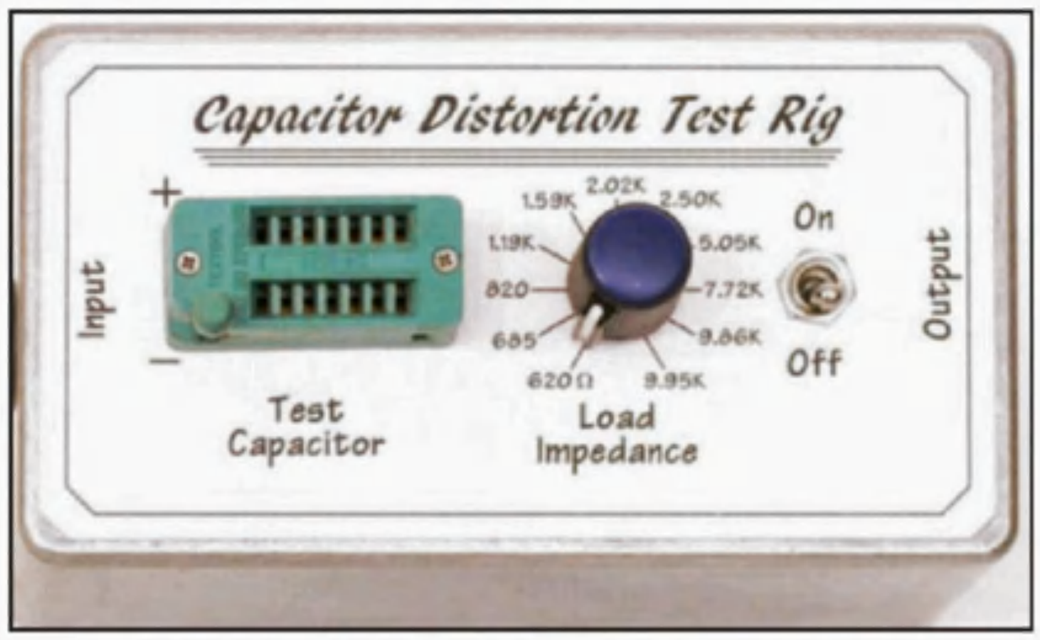
You can use AC on that tube's heater, but you will need to elevate the entire heater string by about one-third of the B+ voltage applied to the driver. An alternative is to have separate heater windings for the driver and the output tubes. The latter do not need to be elevated but there is no harm in doing it.

In my designs for larger power amplifiers, I do just that. In the new design, I changed the driver tube from the type 5751 to ECC802S. The loss in gain was not an issue when used as a power amplifier, but could be a concern for use with low output or passive preamplifiers with insufficient gain. The ECC802S sounds better, in my opinion, and there is

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a wider selection of tubes that can be used. I like the sound of the JJ Tesla ones, but feel free to try any 12AU7 variant. Should you actually need the extra gain, you can directly substitute type 5751 tubes and with slightly worse signal-to-noise 12AX7s or any of its variants.

Design Features

The design is based on the original amplifiers I built using large power tubes back in 2006. I was on a power trip and was certain that nothing short of KT77s and KT88s was worth the effort. The ensuing amplifiers were actually quite good and subsequently became commercial products. Still, they were overkill for many uses and generated a lot of heat.

The Mini Blocks have a similar configuration that is downsized from the big amplifiers. They use a shunt regulated push-pull (SRPP, also called an actively loaded triode) gain stage to feed a self-inverting Class A push-pull output stage. The output stage is restricted to Class A operation only by use of a constant current source (CCS) in the cathodes. It eliminates the need to set actual bias on the tubes but does require balancing the idle current between the tubes.

The ungrounded cathodes and CCS provide the signal source of drive for the "slave" tube, which is the one that does not get signal from the driver stage. The "slave" tube is driven by the signal between the grid and cathode. The AC component is effectively "ground" referenced by the capacitor between the grid and the ground with the cathode elevated by the CCS. The signal is developed between them and is out of phase with the one applied directly to the other tube (see **Figure 1**). Observant individuals will also note in this configuration that I used small (5 to 8) volts of positive voltage applied to the output tube grids. This would seem contrary to usual practice and would cause problems for most amplifiers. Here, it elevates the cathode voltages by a like amount. I did this because of the CCS, which nicely handles this increase.

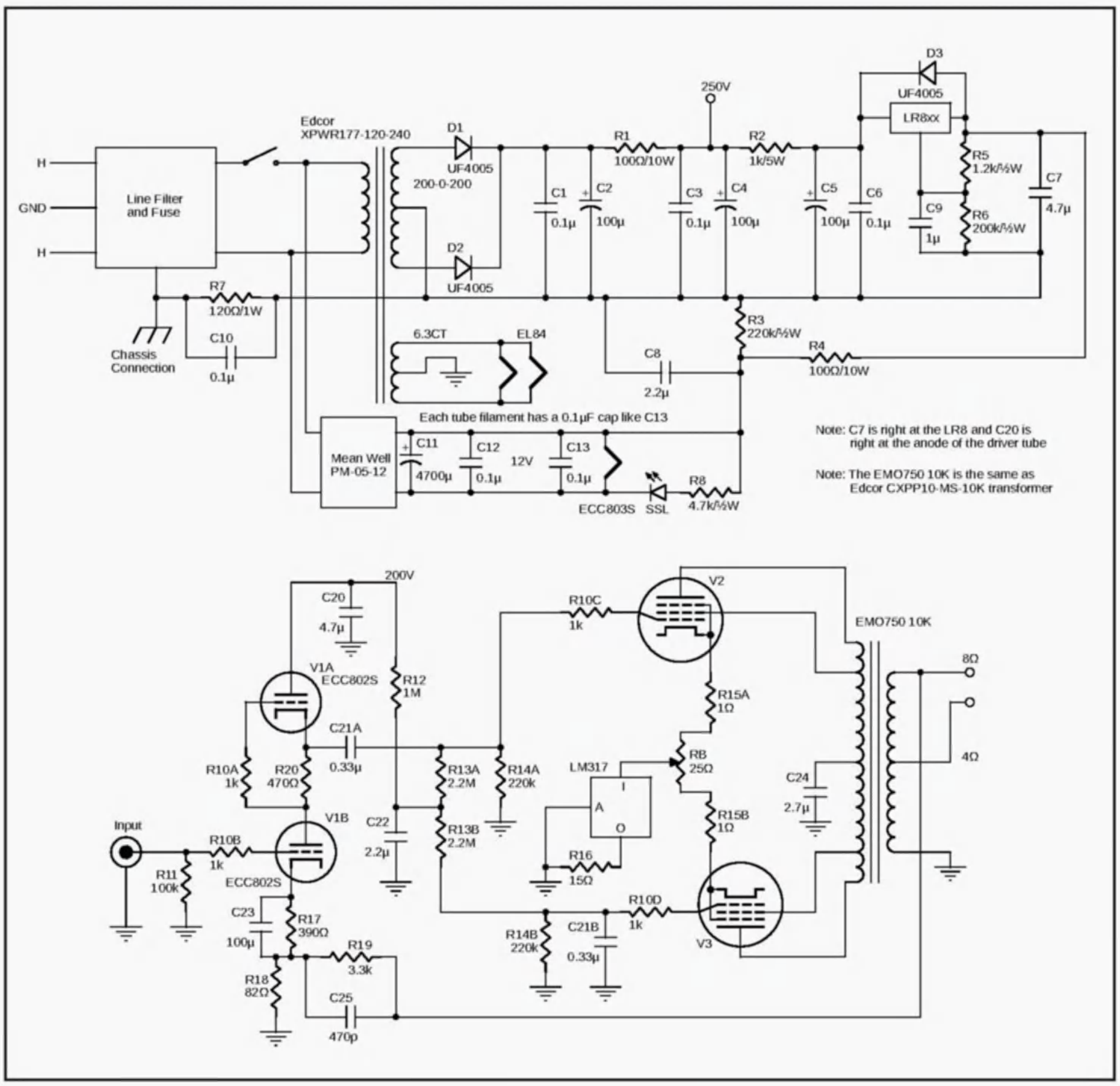


Figure 1: This is the most recent EL84's consolidated circuit.

LM317s used for the CCS become nonlinear below about 4 V. So the increase on the grids shifts the minimum voltage across the LM317 by about 5 V to 8 V and keeps them out of the nonlinear region. Another IC solid-state regulator is used in the high voltage supply. It is used not so much to regulate the voltage, which it does, but to filter the voltage to the driver stage.

The type of LR8XX that I used is rated for about 450 V and, depending on the variant used, has a dissipation of between 750 mW and 2.5 W. They do have some quirks but are great and inexpensive devices. If you short one out even for a fraction of a second, it will fail immediately. If it senses too much dissipation it will fall back to a safe mode and only fully removing the power will restore it to operation. They do not require heatsinks.

The LM317s do need heatsinking, but with suitable insulators you can normally use the chassis like I did in these amplifiers. Remember that the tab on LM317s cannot be grounded. The amplifier, like all of my similar designs, uses a small amount (less than 3 dB) of negative feedback. It is tailored to function primarily at way above audio frequencies. Modeling indicates that these amplifiers, because of the very wide band design, could oscillate at about 70 kHz if there is a really odd speaker/crossover and some external excitation. The negative feedback (NFB) restricts the response to the 35 kHz range. You can make it switchable, if desired, and probably about 99.99% of the time it will be just fine. I have not heard of any that had problems, but there is always a first time.

Construction

You can use point-to-point like I did with perf boards for some components, or PCBs, or any combination desired. Turret boards work fine as well. Layout is not critical, however, good wiring practices should be employed. Even though the gain is low, shielded input cables should be used. Remember to ground only one end of the shield. You should move the driver tube and associated circuitry away from the power transformer as well to avoid hum pickup. A fully enclosed chassis is also recommended to prevent accidental contact with dangerous voltages and to enhance electromagnetic interference (EMI) shielding (see **Photo 2**).

One concern that sometimes occurs is that coupling capacitors can be sensitive to nearby sources of hum and noise. Some are built with an "outside" foil layer that should be attached to the side with the lowest impedance to either the B+ or ground. However, most capacitors are no longer marked to indicate this. If you find ones that seem to be sensitive, try swapping the leads.

The ones in my build (Audyn TrueCopper) do not seem to be sensitive in this manner. Some Jantzen Silver Z capacitors I tried did seem sensitive. Russian K40-Y9 capacitors with the bare metal cases are sensitive and need to have their cases attached to a signal ground. You can actually solder on their cases. A clamp would do as well.

The chassis is isolated from the signal ground for both safety and EMI reasons. A single connection using a type X2 capacitor with a parallel resistor goes between the signal ground and chassis. I tend to use

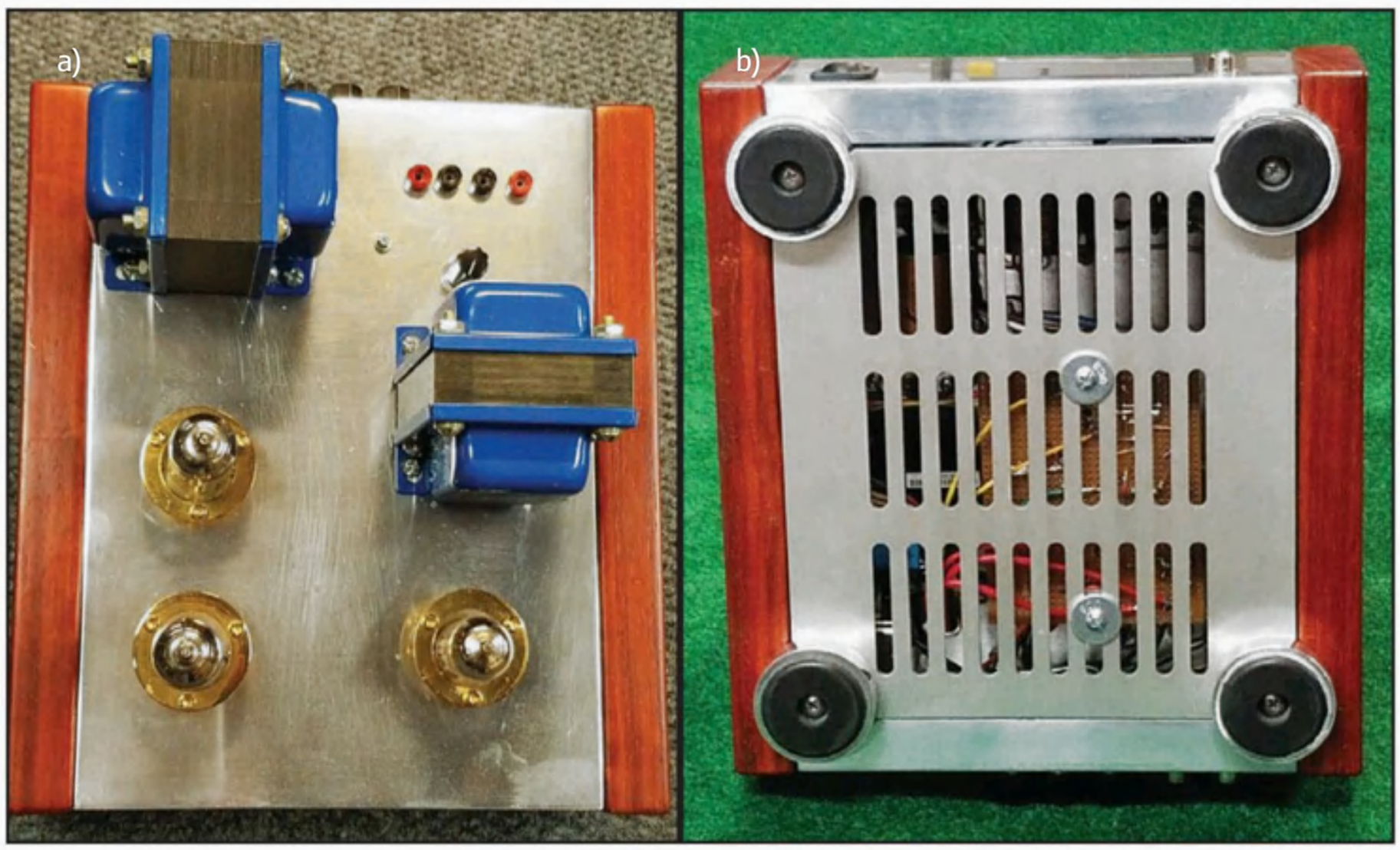


Photo 2: A fully enclosed chassis is recommended to prevent accidental contact with dangerous voltages and to enhance electromagnetic interference shielding. Shown are the top (a) and the bottom (b) of the EL84 tube amplifier.

a star grounding scheme with sub-stars. I find that a single-point signal ground at the input jacks works best for all the "stars" to come together. You should make a single connection from the power supply section and separate ones from the active sections to it. Do not route any ground carrying power (e.g., from the filters and power transformer) through the same wire as one that is used as a signal ground. You can use a ground buss if it is large enough as long as the signal connections and power ones feed it separately. Number 12 wire is usually suitable for this.

One build consideration is that the output transformers are used to invert the signal. If you use the specified Edcor ones this means that the brown wires go to the "driven" tube and the blue ones to the "slave." The specified output transformers were designed for me by Edcor for this particular application (see **Photo 3**). You certainly can use others but the results may differ a bit.

Initial Setup

Initial startup and operation are simple. Many first-time builders think that you must test the voltages without the tubes in place and they get confused by the readings. They will be well above the ones when the amplifiers are operating because there is no current being used by the tubes. It is OK to do that sort of testing as long as you are aware of the issue and, more importantly, that the filter capacitors you use are rated for the higher voltage.

I suspect more than one person has fried a filter capacitor that was only rated for the operational voltage and not the excess when there is no current being used. The differences between the no load and full load voltages in these amplifiers is about 30%. If you do that sort of test and all is about as expected then it is time to insert the tubes. At this point, neither an input nor speaker load is essential.

With this design, I like to set my first test point at the junction of the balance control to the ground. If everything is working well, it ought to be in the 10 VDC to 25 VDC range. If it is much higher or much lower then there is a wiring error. If it is OK, then measure the millivolts DC across the test points. These are the 1 Ω resistors in the cathode circuits. Do not measure to the ground or between the alternate tubes as the reading will not be useful.

The voltages across the 1 Ω resistors should be about equal and around 40 mV to 45 mV. If not, the balance control should be adjusted. If you cannot balance them, then try switching the tube positions. If it still will not balance, then the tubes are too badly mismatched. A matched set is not usually needed, but I find it simplifies things. If one tube shows a nearly zero reading across its test points, then there is a wiring fault. This is often in the grid circuit. Either tube might have a poor grid to ground connection. The one with the poor connection will then float high and cause it to hog the current. The balance should be checked after an hour or so and reset as required.

Periodic checking is important and I find that after the first few hours that a 3 to 6 month interval is about right. If the tubes were well matched then the changes will likely be very small. Tube aging of mismatched tubes will usually result in larger deviations. If all is well at this point, it is time to hook up the speakers—if you have not already done so. There should be no hum or noise when no signal is fed in. If there is, you need to hunt for either a poor ground or a component placed too near a source of hum. Typically these amplifiers have a signal-to-noise ratio (SNR) in excess of 80 dB at 1 W.

Operation

The amplifiers require the use of a preamplifier in most systems. You can replace the input 100 kΩ resistor with a like-value volume control and use them without a preamplifier if you have a signal source that will provide sufficient output. A preamplifier with an output impedance that is less than about 50 kΩ is optimum, however, others may be suitable. Ones that have an output impedance over that may alter the frequency response.

When provided a good input signal, the sound should be clean and full range. If it sounds sort of

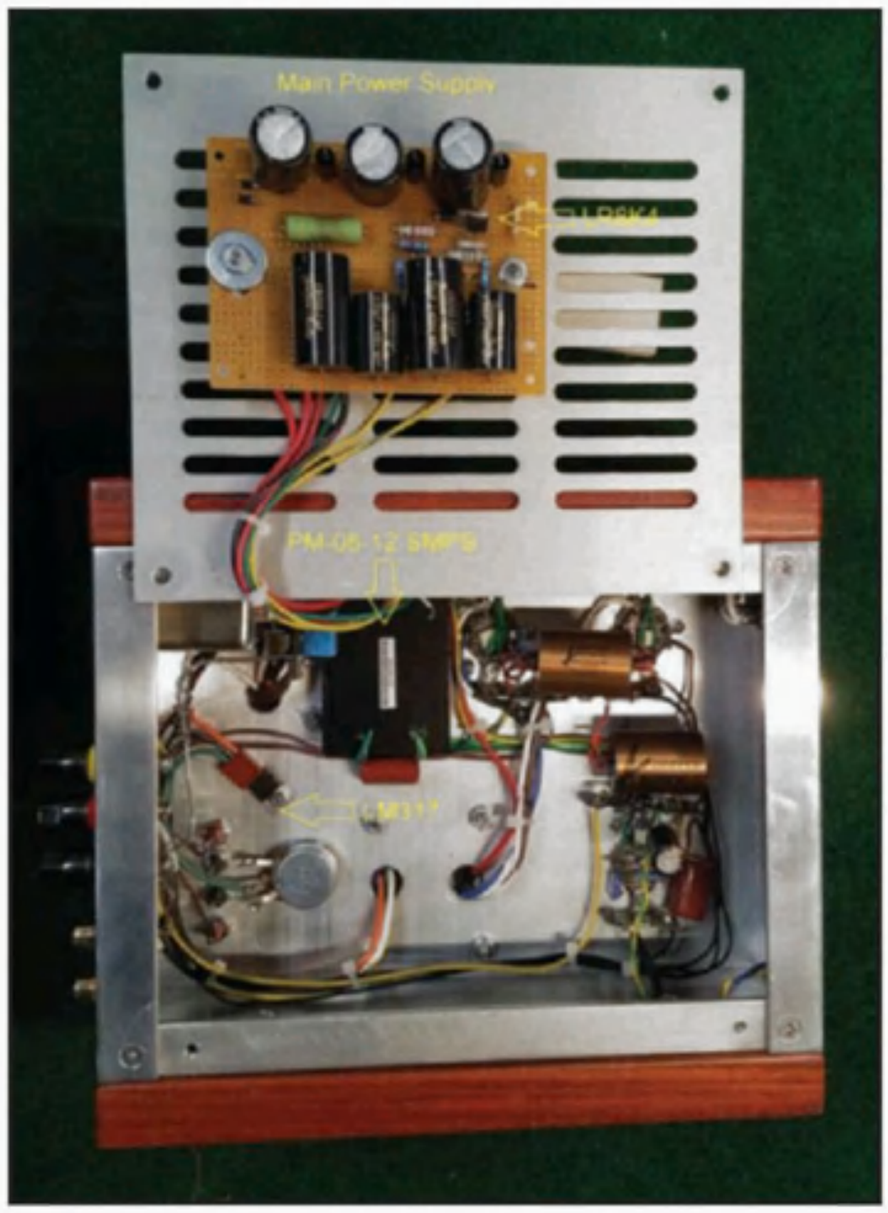


Photo 3: If you use the specified Edcor output transformers, the brown wires go to the "driven" tube and the blue ones to the "slave." The specified output transformers were designed for me by Edcor for this particular application.

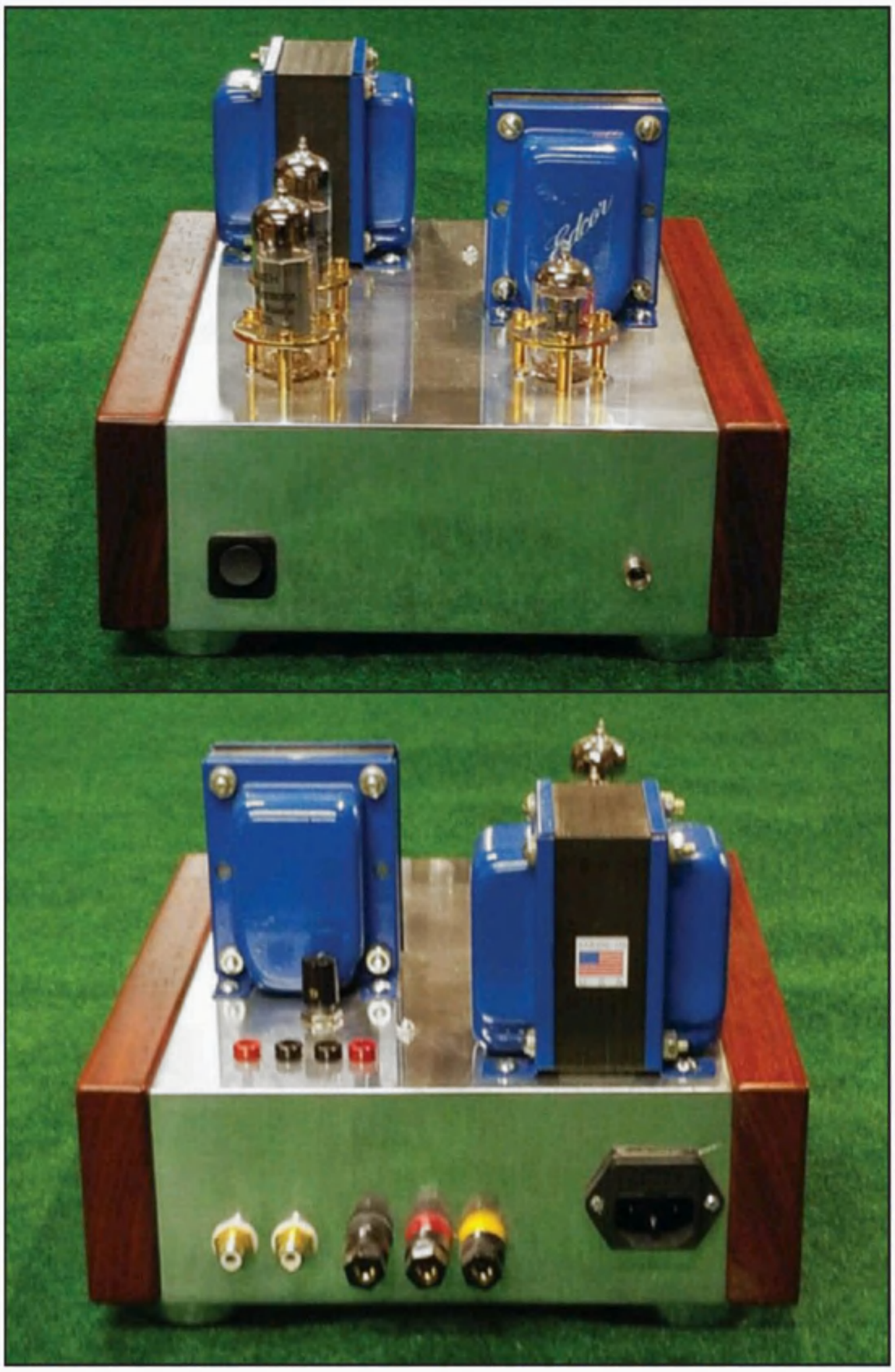


Photo 4: This shows the front (a) and back (b) of my updated 2020 version of the EL84 Mini Block Tube Amplifier.

shrill and lacks bass, then it is likely that the output transformer primary connections are incorrect. The transformer is used to invert the signal so that the overall amplifier behavior is not inverting. If this problem is evident, you can check it by disconnecting the negative feedback and see if it sounds better. If so, you need to correct the wiring. While not obvious, you can also switch which tube is used as the slave by changing the grid circuits.

Components

Small components such as resistors and capacitors can be selected by the builder. The same is true for the filter capacitors. The coupling capacitors have a major effect on the sound and should be carefully selected. They need not be premium ones, but you should choose ones that sound appealing. The output transformers I used are from EDCOR. They may be difficult or expensive to obtain for non-US builders. Others can be used, but may or may not be sonically equal. The power transformers can be generic as long as they are about the same ratings.

Builders can use AC heaters on the driver tubes if desired, with a slight signal-to-noise penalty. It is also possible to rectify a filament winding and provide DC for the heaters as long as you do not ground the circuit and do elevate it by the necessary amount to prevent heater to cathode failure of the driver tubes. The balance control only dissipates about 0.25 W maximum so it need not be huge. A 1 W one is fine. Wire wound, carbon, or plastic ones can be used but be aware, since it handles a DC current, it will need to be exercised from time to time to prevent an oxidized spot. The normal 3 to 6 month balance checking interval is fine.

Listening

This part is always subject to individual opinions. I find these amplifiers (see **Photo 4**) really nice with a clean full-range sound. Using the components I did makes the response flat to 10 Hz and starts to roll off (because of the NFB stability components) at about 35 kHz.

The one drawback that some folks will encounter is that these are 5 W amplifiers and suitable for moderately sensitive speakers. They will not have sufficient power to drive speakers that require huge amounts of power. I use them routinely with MartinLogan electrostatics that have a sensitivity of about 91 dB/W in a room that is about 4 m wide, 6 m deep, and 3 m tall. For anything but seriously loud music, the amplifiers do just fine.

About the Author

Bruce Heran lives in Sierra Vista, AZ, and is an avid DIYer. He has been involved in electronics for 59 years. He built his first project—a one-transistor amplifier—when he was 13. He has a BS in Biology and studied electrical engineering for two years. Electronics has always been his hobby. In May 2011, he retired from his work as a project manager of a logistics contract with the Federal government. Now he is co-owner of Oddwatt Audio (http://oddwattaudio.com), an electronics company that specializes in vacuum-tube audio components. He is vice president of design and support, and all of the company's products are kits and assembled audio equipment based on his personal designs. He is an avid collector of vinyl records and vintage turntables.

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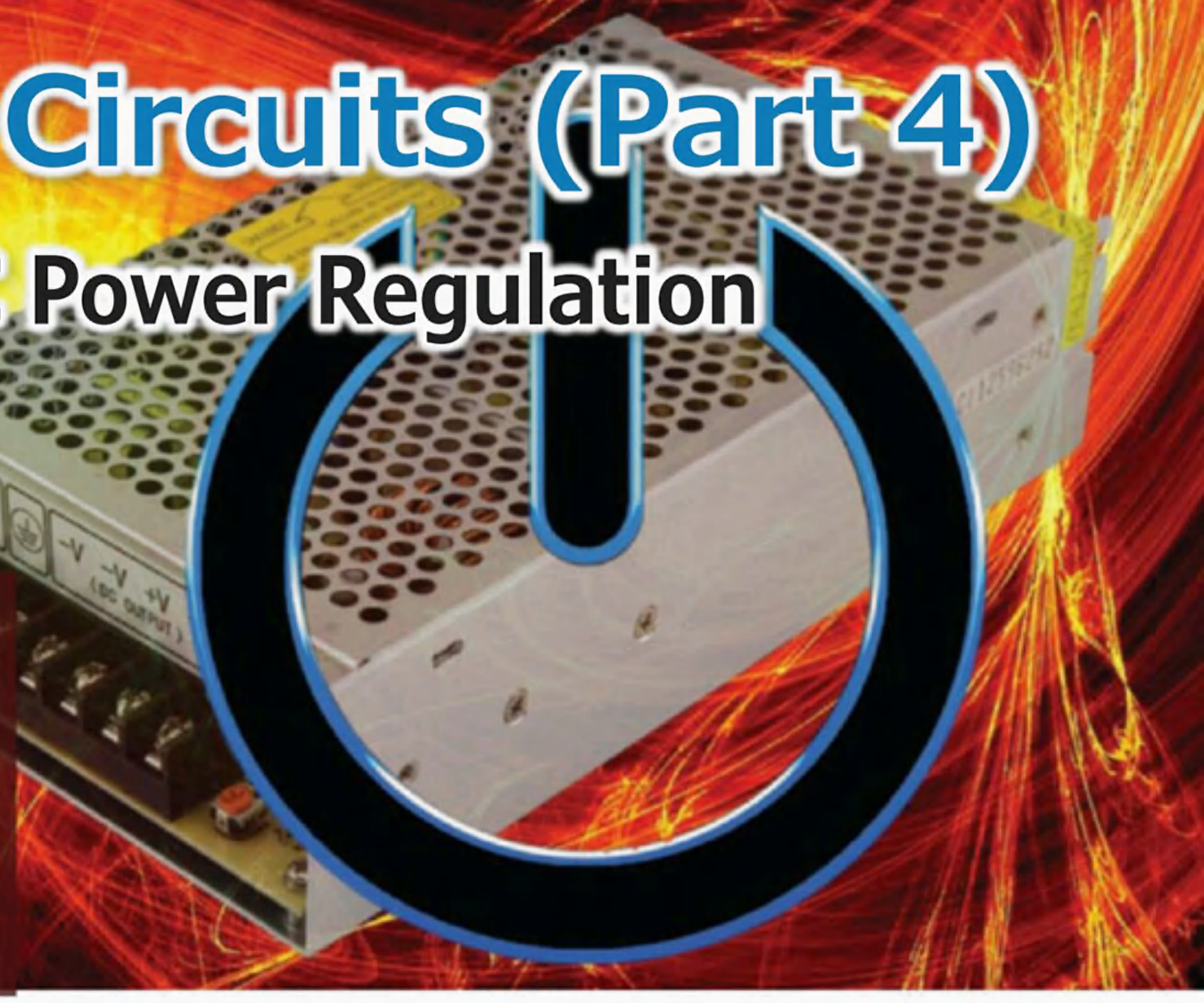
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Powering Your Circuits (Part 4)

AC Rectification and DC Power Regulation

By
Ed Simon
(United States)

In this article series, Ed Simon continues to detail ways to power circuits. After discussing battery power and transitioning to AC power supplies, this article continues to explore AC waveforms and looks at full-wave bridge (FWB) circuits and other forms of DC power regulation.



In the previous article, we were looking at lessening power supply noise and we discussed what happens when we increase the filter capacitor size. At the peak, the AC line is at maximum voltage, current is at a maximum, and so are the internal losses. After the current peak, the AC line drops in voltage; the secondary drops from this, but at the same time rises due to the lesser load current and smaller losses. This extends the charging time.

For those readers really looking closely at the top of the AC waveform when it just begins to drop, the capacitor would normally discharge faster than this but the AC waveform still holds it up. This occurs for about 2.5° from the peak.

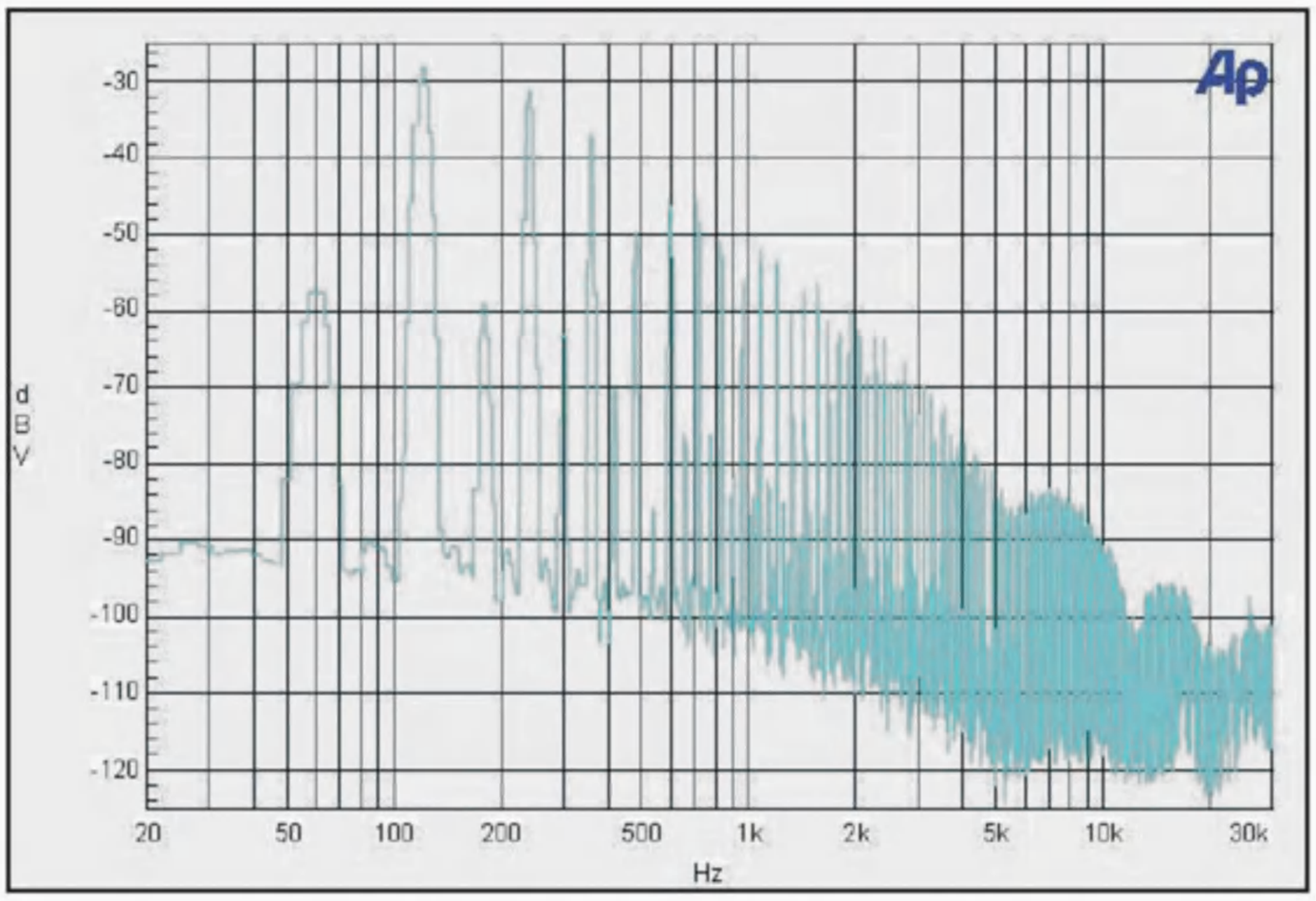


Figure 1: The FFT of the current with the larger capacitor shows more 60 Hz and more 240 Hz draw.

The extended charging time and change in pulse shape has some effect on the noise created. If you look at the Fast Fourier Transform (FFT) of the current with the larger capacitor (see **Figure 1**), you will see more 60 Hz and more 240 Hz draw. What is interesting is the increase in noise centered around 7,000 Hz.

Real capacitors are not perfect, as the frequency increases the imperfections cause the reactance to be a bit higher. Larger value capacitors will have a lower high-frequency limit than smaller units of similar construction. So a single large-filter capacitor may require some help. This is often accomplished by adding a second much smaller capacitor in parallel to the main filter capacitor.

One other item of note is that when a capacitor with a DC voltage across it is moved or vibrated, any internal plate movement will cause a slight change in capacitance, which will show up as a current. In short, the capacitor can act as a microphone. Some capacitors (mostly ceramic) actually have a piezo-electric dielectric and can generate currents from vibrations.

Now you might have wondered why for a 50 Ω load or a current draw of 0.32 A use a 12 A transformer? A transformer with a lower current rating would work, the main result being lower charging current available.

Figure 2 shows the charging current with the same load on a smaller transformer. This particular smaller transformer is a dual primary, dual secondary "O" core flat pack PC-mounting type. It is rated for 16 VCT and 800 mA. As you can see, the charging time has increased a bit even though with lower voltage less total charge needs to be replaced in the capacitor filter.

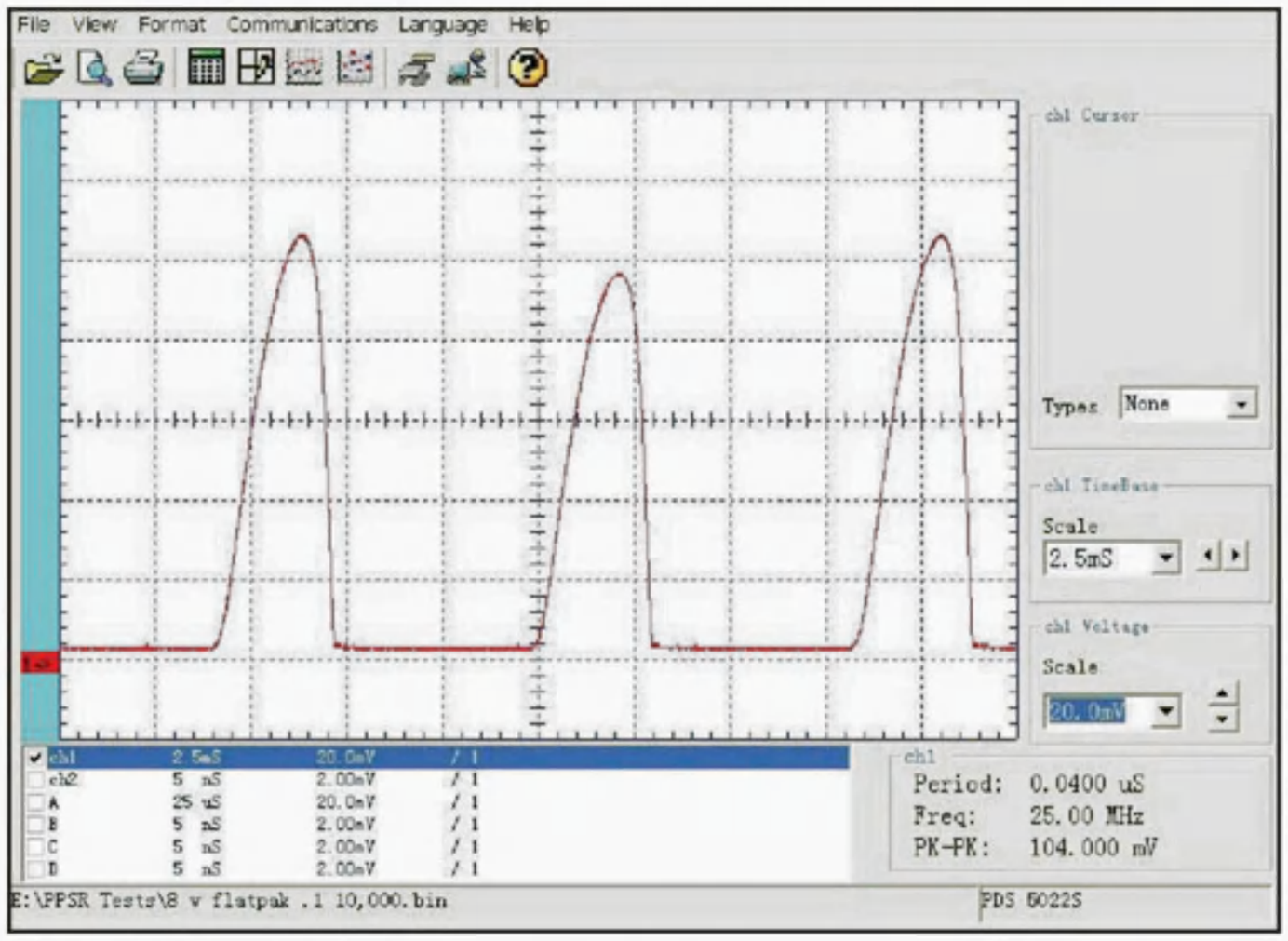


Figure 2: This particular smaller transformer is a dual primary, dual secondary "O" core flat pack PC-mounting type.

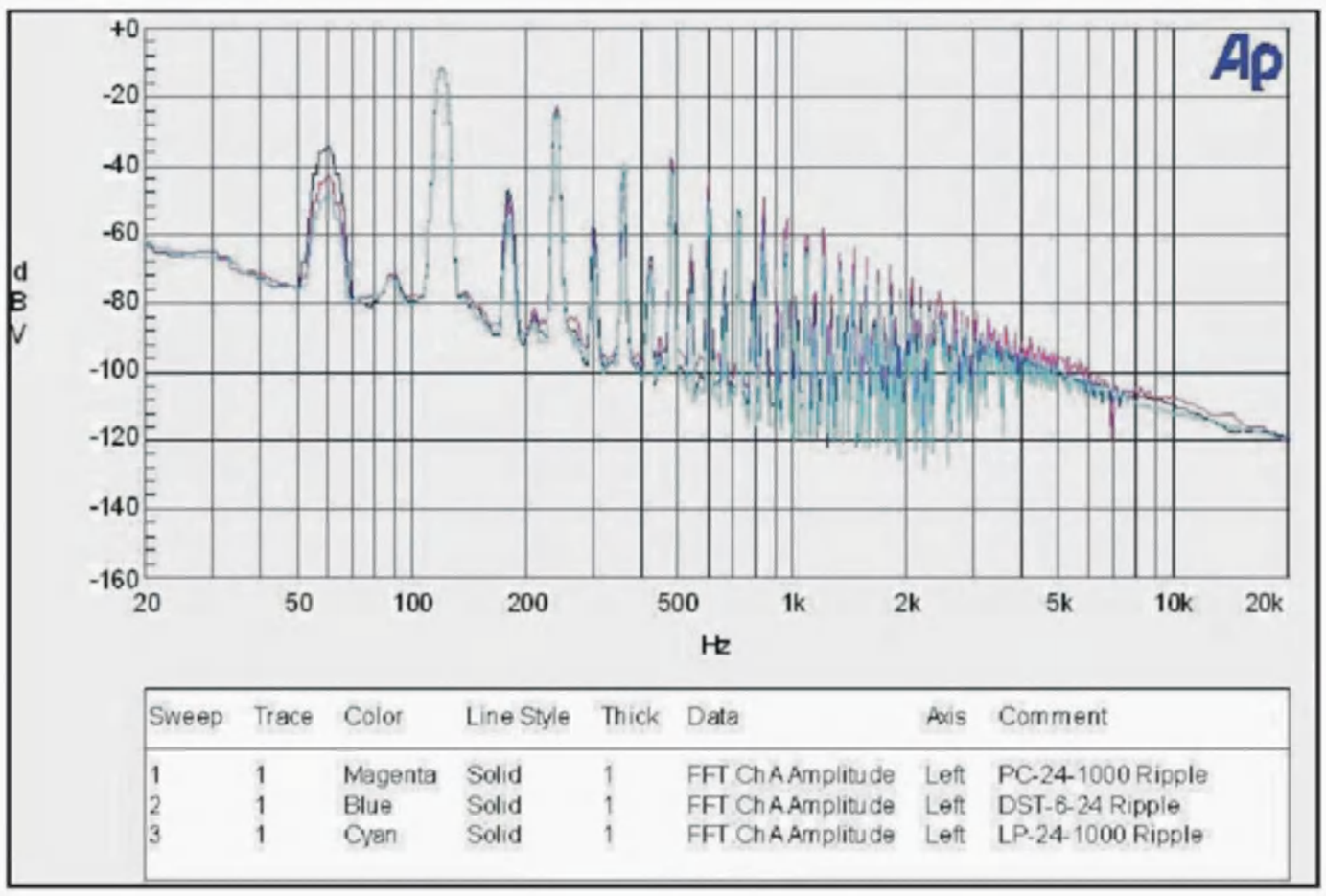


Figure 3: With a smaller transformer, the noise produced with the same load on goes up a bit. This graph shows the ripple noise from three different, but similar capacity transformers.

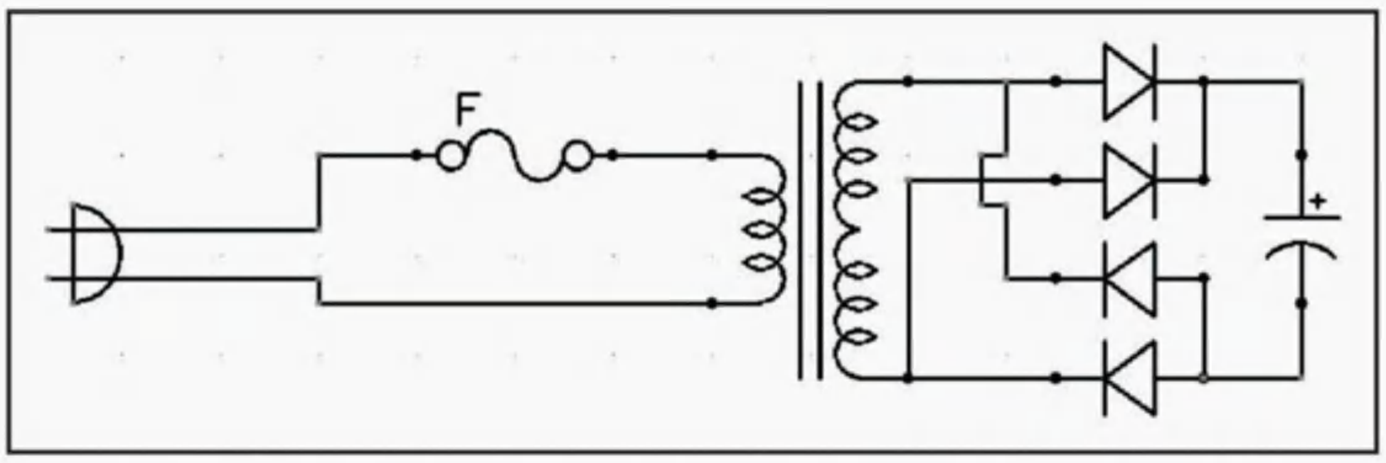


Figure 4: This Full Wave Bridge (FWB) circuit has two paths for the current to reach and charge the capacitor.

What is more interesting is that the charging pulses are not the same height. As the only part that has been changed is the transformer, this indicates the transformer does not have truly balanced secondary windings! There is about an 8% difference in charging current.

Now this uneven loading of the transformer core most likely will cause more even order harmonic noise in the power supply! (The clever readers will go back and review the chassis flux curves and note that this transformer has a different harmonic signature in those curves!)

This transformer also has a split primary. If the primaries are mismatched, there will be a bit more distortion and a small reduction in maximum current when used in parallel.

With a smaller transformer, the noise produced by the same load goes up a bit. **Figure 3** shows the ripple noise from three different but similar capacity transformers.

A first approach to reduce this is to use an active voltage regulator. A very brief look at a few single-chip regulators show that they are very good at low frequencies, but are not constant with frequency.

Full-Wave Bridge Circuits

So far, we have mostly looked at center-tapped single DC voltage power supplies. Now, the disadvantage of the center-tapped single voltage supply is that the secondary winding needs twice as many turns than using a single winding. As diodes dropped in price, it became cheaper to use four diodes in a circuit called a bridge to load the transformer on both half cycles. This used four diodes and only needed one winding.

Now the Full Wave Bridge (FWB) circuit, shown in **Figure 4**, has two paths for the current to reach and charge the capacitor. One path uses the top and bottom diodes. When the AC reverses it use the middle ones. This means that there are two diodes

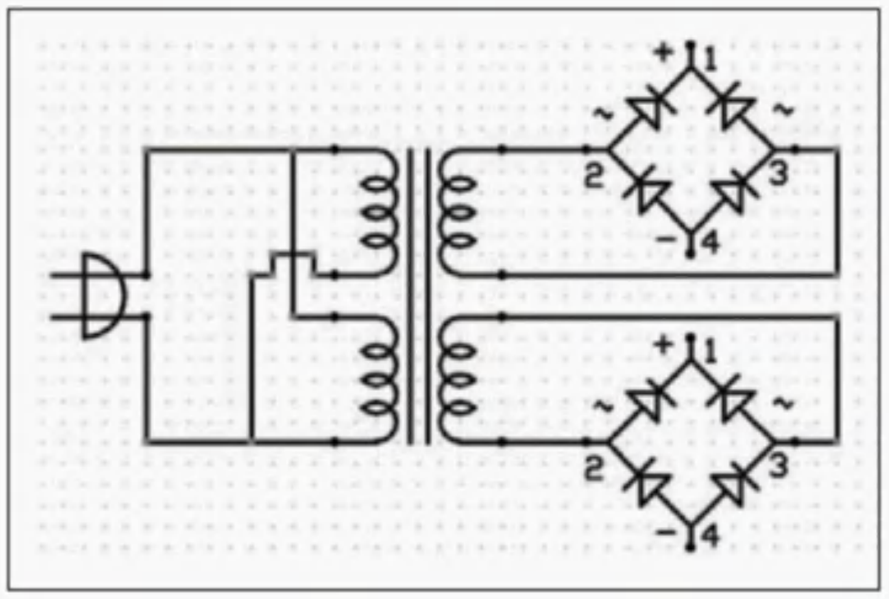
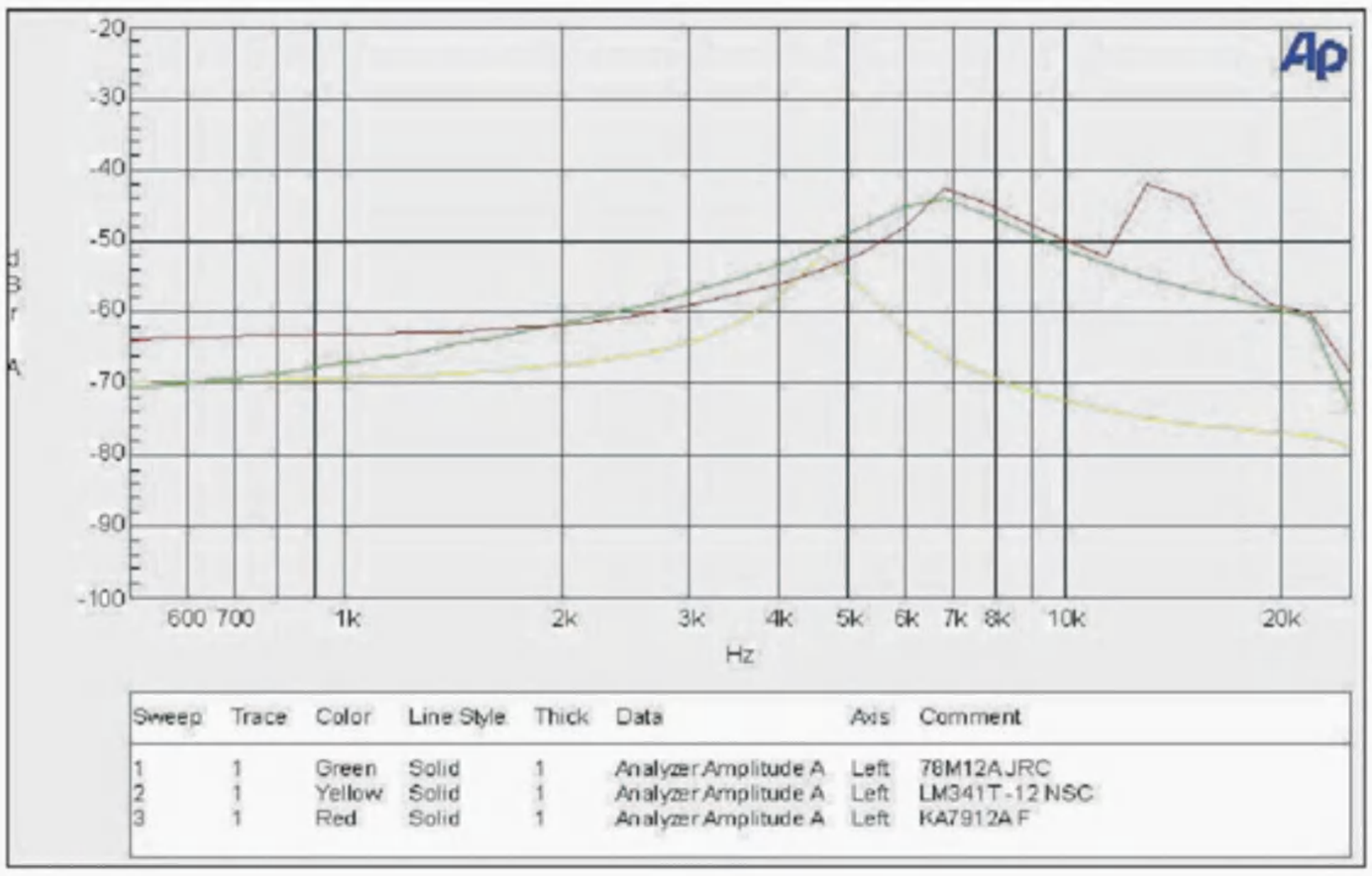
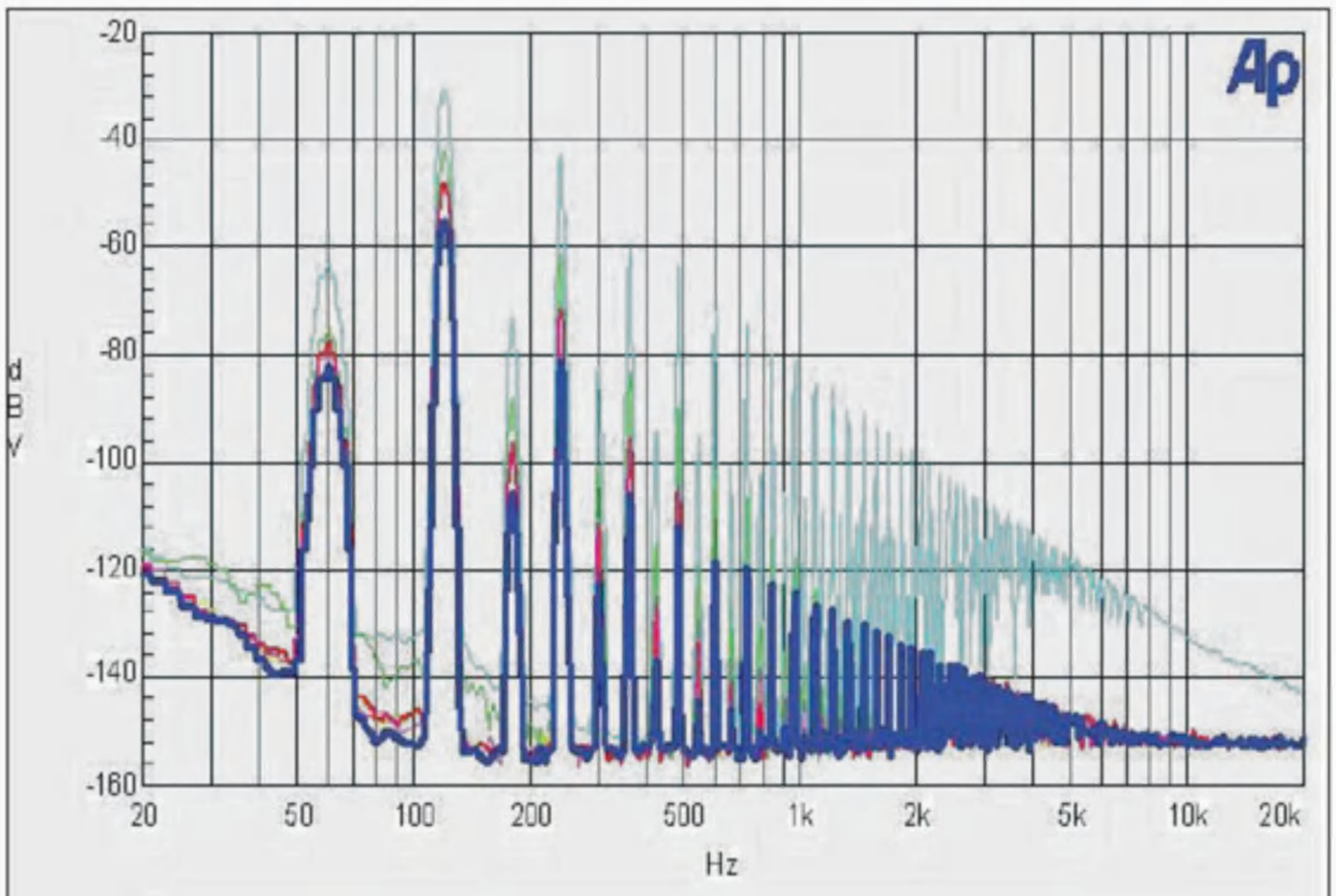


Figure 5: Using two bridge rectifiers will assure that when a secondary coil swings either way with the AC voltage that there is the same current drawn.



Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment
1	1	Green	Solid	1	Analyzer.Amplitude A	Left	78M12AJRC
2	1	Yellow	Solid	1	Analyzer.Amplitude A	Left	LM341T-12 NSC
3	1	Red	Solid	1	Analyzer.Amplitude A	Left	KA7912AF

Figure 6: This graph shows the results after I hooked up a simple circuit that fed regulators a DC supply voltage with a bit of swept AC added.



Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment
1	1	Cyan	Solid	1	FFT.ChA.Amplitude	Left	18,800 uf
2	1	Green	Solid	1	FFT.ChA.Amplitude	Left	9400 uf 1 ohm 9400 uf
3	1	Yellow	Solid	1	FFT.ChA.Amplitude	Left	9400 14700 14700
4	1	Red	Solid	2	FFT.ChA.Amplitude	Left	4700 19400 14700
5	1	Magenta	Solid	1	FFT.ChA.Amplitude	Left	4700 14700 19400
6	1	Blue	Solid	3	FFT.ChA.Amplitude	Left	4700 14700 14700 14700

Figure 7: The filter capacitance has been increased.



Photo 1: In order to get the actual capacitor into the radial case, I used a longer lead.

in series to charge the capacitor. So the diode loss is doubled. An advantage is that since the diodes are in series the reverse voltage rating required of the diodes is half that of the other circuits.

If you look at Figure 2, you will see the charging current is not symmetric for a dual coil transformer. It turns out there is a method to insure that the current draw is symmetrical even when both secondary coils are slightly different.

Using two bridge rectifiers, as shown in **Figure 5**, will assure that when a secondary coil swings either way with the AC voltage, the same current is drawn. If the sources are placed in series, then the voltage would be higher and both coils would be loaded.

You could also use the dual-coil dual-bridge circuit with the center grounded as a plus and minus rail power supply. If the coils are matched, the raw DC would be the same. If not, there will be a small difference. In some cases, it is advantageous to keep the two supplies apart and only combine them at a common ground point for all of the rest of the circuit. This allows the current used to charge the filter capacitors to be completely separated from any other ground currents in a circuit.

More Regulator Tests

I hooked up a simple circuit that fed regulators a DC supply voltage with a bit of swept AC added. The results of a few tests are shown in **Figure 6**. Most regulator designs sense the output voltage and use feedback to control it. As the frequency increases the gain of the feedback amplifier falls. This results in less regulation. The units shown are common types and, unfortunately for audio use, lose a bit of regulation at the frequencies of greatest hearing acuity!

On the output of the IC regulator under test is another filter capacitor as suggested by the manufacturers and this reduces the ripple and noise pass through at higher frequencies. A lesson here is that regulators have practical limits. (The wise may also evaluate regulators by switching the load current and looking for ringing!)

In low current power supplies, there is another way to reduce noise besides just adding capacitors and regulators. In **Figure 7**, the filter capacitance has been increased. The top curve uses a single capacitor bank of four 4,700 µF capacitors for a total capacitance of 18,800 µF. The next curve is has almost 10 dB lower noise. It uses two 4,700 µF capacitors in parallel feeding a 1 Ω resistor that is again filtered by two more 4,700 µF capacitors, this is known as a CRC filter. So for the same total capacitance there is a 10 dB improvement in noise. As the load is 50 Ω the additional voltage loss from the resistor is 2% (or 0.175 dB).

We can use the same trick by using two resistors and three stages of capacitors. The fourth capacitor in the set is shown in all three possible positions and pretty much makes no difference where it is located.

Finally we use the four capacitors and three resistors for a total voltage loss of less than 6% (0.51 dB). This has reduced the power supply noise by more than 25 dB. The ripple is now just a few millivolts. Unlike a regulator as you stack more RC sections you continue to drop the noise. The disadvantage is the increase in voltage drop and lack of precise output voltage.

Now you may have noted there is no boost in noise in the ranges shown from capacitor resonances. It is not just that smaller value capacitors were used but also the type of capacitor. Here, we are using radial capacitors. Before the filter was an axial capacitor. In order to get the axial capacitor into the radial case, I used a longer lead (see **Photo 1**). This adds to the Equivalent Series Resistance (ESR) and Equivalent Series Inductance (ESL, using the symbol L for inductance) of this style of construction. This results in lower resonant frequencies. Note that film capacitors are not placed in metal cases and do not use the same limiting construction.

Another issue with capacitors is that because they have internal wiring there is some loss due to the internal inductance. This is called the ESL. Larger values of capacitance generally have higher inductance. This causes the capacitor to form an LC resonance at some frequency, which means every capacitor has a limited useful filtering frequency range.

There is also some internal resistance. This is called ESR. The lower this is, the better the capacitor works as a filter generally. Capacitors also heat up from this internal resistance. If they get too hot, they often fail dramatically with a nice big bang! To prevent this from occurring, there is not only a voltage limit for any given capacitor but also a current limit. The formula: C dv/dt = i is useful for this calculation, if you know the ripple voltage, capacitance, and charge time. The charge time is 1/f or 0.00833 for a 120 Hz full wave bridge.

The other operating issue is temperature. Freezing is bad for most electrolytic capacitors as is running in too hot an enclosure. So capacitors have a temperature rating. However, picking a higher temperature-rated capacitor than needed is usually not a good idea as the higher temperature-rated units often have higher ESR.

The final issue of ESR is that like any resistor it acts as a noise source! This is similar to the internal loss of batteries. So picking a capacitor is not just capacitance and voltage, but also ripple current, temperature, ESR, and often, size and mounting considerations.

Next month, we will tackle the topic of mains noise filtering.

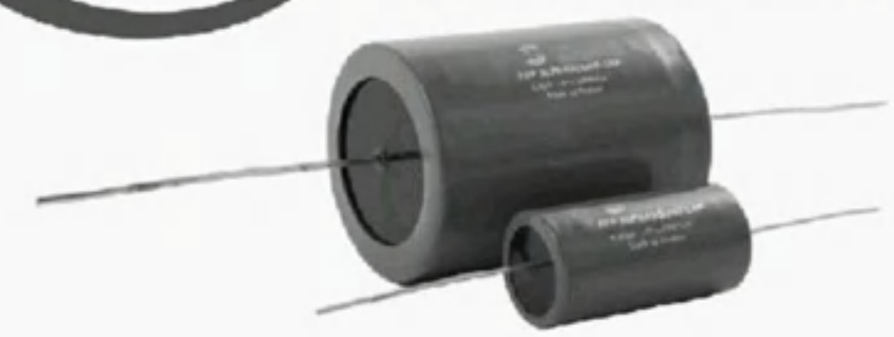
About the Author

Ed Simon has a BSEE from Carnegie-Mellon University. He was employed by Cameradio, a regional electronics parts distributor from 1973-1975. In 1975, he started Edward Simon & Co. He has published more than 40 articles and taught courses for the NSCA, Carnegie-Mellon University and the University of Pittsburgh among others. His company specializes in sound and communication systems, specializing in systems for public facilities, houses of worship, and allied engineering. The business has completed more than 400 church installations, and 150 other installations including shopping malls, teleconferencing facilities, performing arts centers, television studios, and radio stations.

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Adventures in Guitar Amplifier Repair

In the more than 50 years that Richard Honeycutt has been repairing hollow-state guitar amplifiers, he has learned that certain models exhibit certain specific failure modes, and if you get to know these modes, you can dramatically cut troubleshooting time. In this article, he details some “lessons” in amp repair that he has learned over the years.

Photo 1: The 7C5 beam power tube was the loctal equivalent of a 6V6GT.

By
Richard Honeycutt
(United States)

My first electric guitar and amplifier were bought as a package from a friend, for $35, in 1964. The guitar, whose brand name I no longer remember, was made of a rather soft wood, painted gold, and had no truss rod. The neck felt a lot like a baseball bat. The pickup was single-coil. The amp, whose brand name I think was no longer legible, used loctal tubes: a single-ended 7C5 beam power tube (see **Photo 1**), with a 7F7 dual triode as a preamp and a voltage amp. The guitar amp put out about 5 W into an 8” Magnavox speaker. Problem was, it hummed.

Aside from jokes about the amp “not knowing the words,” this was annoying, to say the least. My early studies in electronics repair made me suspect the filter capacitor. Having no way to test the capacitor at that time, I tried replacing it, but there was no improvement.

Digging further into my books, I found that poor ground connections could be responsible for the hum. These could occur in home-brew or poorly repaired equipment in which acid-core solder had been used—not an uncommon problem in equipment of the 1940s and 1950s. The acid would corrode the conductors within the solder joint. Careful inspection revealed that the chassis had defects in the zinc plating, but running heavy wires among all the ground points (soldered, of course, with rosin-core solder) did not help.

In later years, I learned about “star” grounding (see **Figure 1**), which eliminates common grounding impedance issues that can produce hum, and I realized that redoing the grounding could perhaps have fixed the problem. Since the amp was used when I got it, I don’t know whether it had always hummed or not. Common grounding impedance issues (several ground points connected through a common wire to chassis ground) would have made it hum even when new. Eventually, I noticed that the chassis hole where the input jack was mounted was corroded, making a poor ground connection. This repair taught me to always look first for this sort of error, especially if there was hum in an amp that

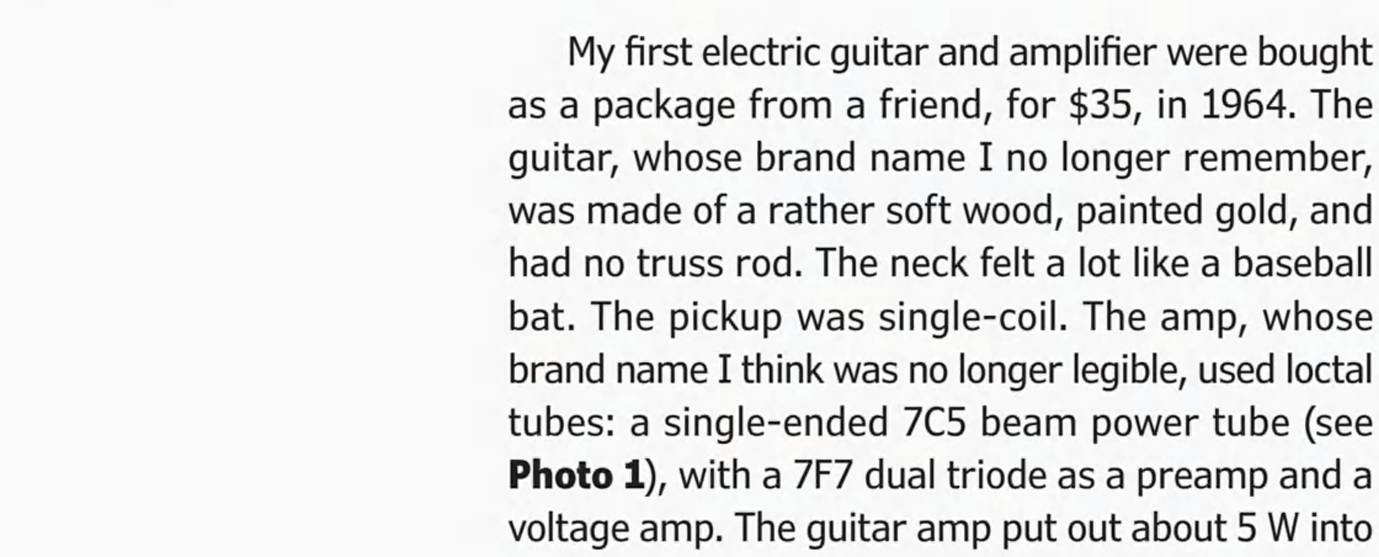


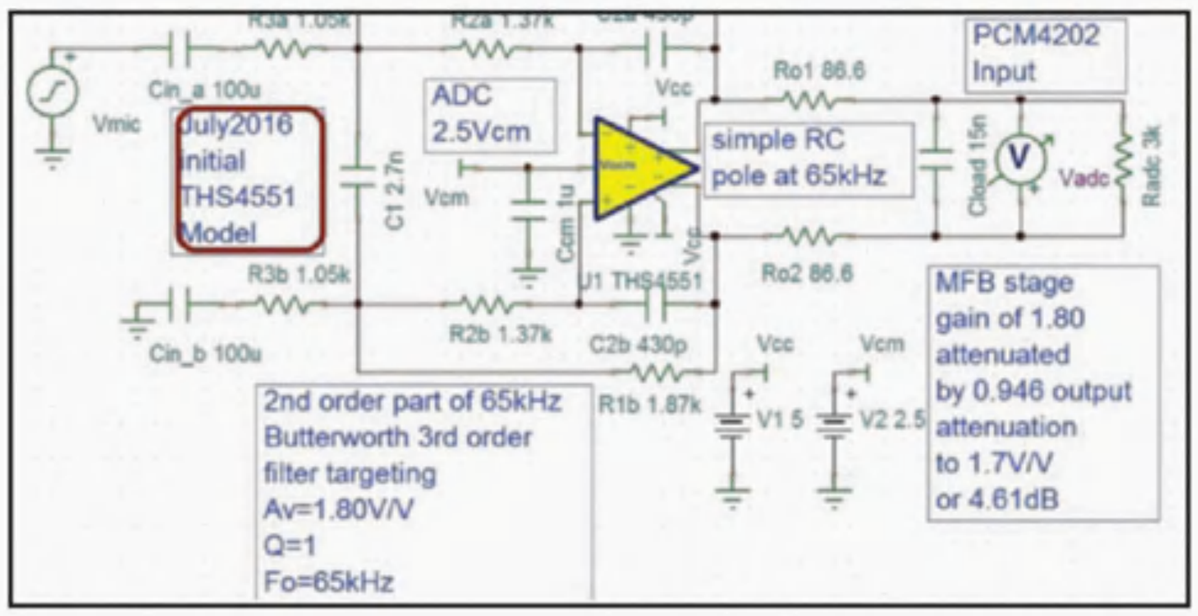
Figure 1: Star grounding helps prevent hum.

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Glass Audio Special

44 Repurposing an Old Preamp for Line-Level Duty

By Bill Reeve

Bill Reeve builds a tube preamplifier to position at the input to his home power amplifier, eliminating the need to adjust his 1972 McIntosh MX10 tuner and 240 power amplifier's volume high when he wants to fill his home with music.

50 Mini Blocks—Small Vacuum Tube Power Amplifiers

An Evolutionary Project Spanning 11 Years

By Bruce Heran

More than a decade ago, Bruce Heran designed and built a small tube-based power amplifier for his office. It worked far better than he expected and was in use with only small updates for seven years. This latest incarnation of that amplifier is actually a pair of mini blocks as they better suit his current needs and include a number of modifications that enhance their performance.

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had been stored in a basement or near the ocean, or was of unknown history.

Common Amplifier Issues

Since the first amp never sounded very good, I replaced it with a Silvertone Twin-Twelve "50 W" amp (see **Photo 2**) made by Valco. As you can see from the circuit in **Figure 2**, the power supply is a voltage-doubler type, so C26 and C28 (two of the four red tubular electrolytics shown in **Photo 3**) do not have the negative plates grounded. If you insert them with the negative plate grounded, or with the polarity reversed, you will blow a fuse and probably damage your new capacitor.

Soon after graduating from college, I began repairing guitar amps professionally part-time. One amp that I saw frequently (because so many were sold) was the Fender Twin Reverb, whose schematic is shown in **Figure 3**.

Shorted rectifier diodes were a common problem with these, especially in the summer, when thunderstorms were common, creating voltage surges on the power lines. You'll notice that the full-wave rectifier was made up of two sets of three solid-state diodes in series. By the 1970s, 1 A, 1 kV diodes were commonly available at a reasonable price, but when the earliest Twins with solid-state rectifiers were made, high-voltage diodes were harder to come by. Connecting three 500 V diodes in series would give the equivalent of a 1.5 kV diode.



Photo 2: This Valco-manufactured 50 W Twin-Twelve guitar amp was very popular in the 1960s.

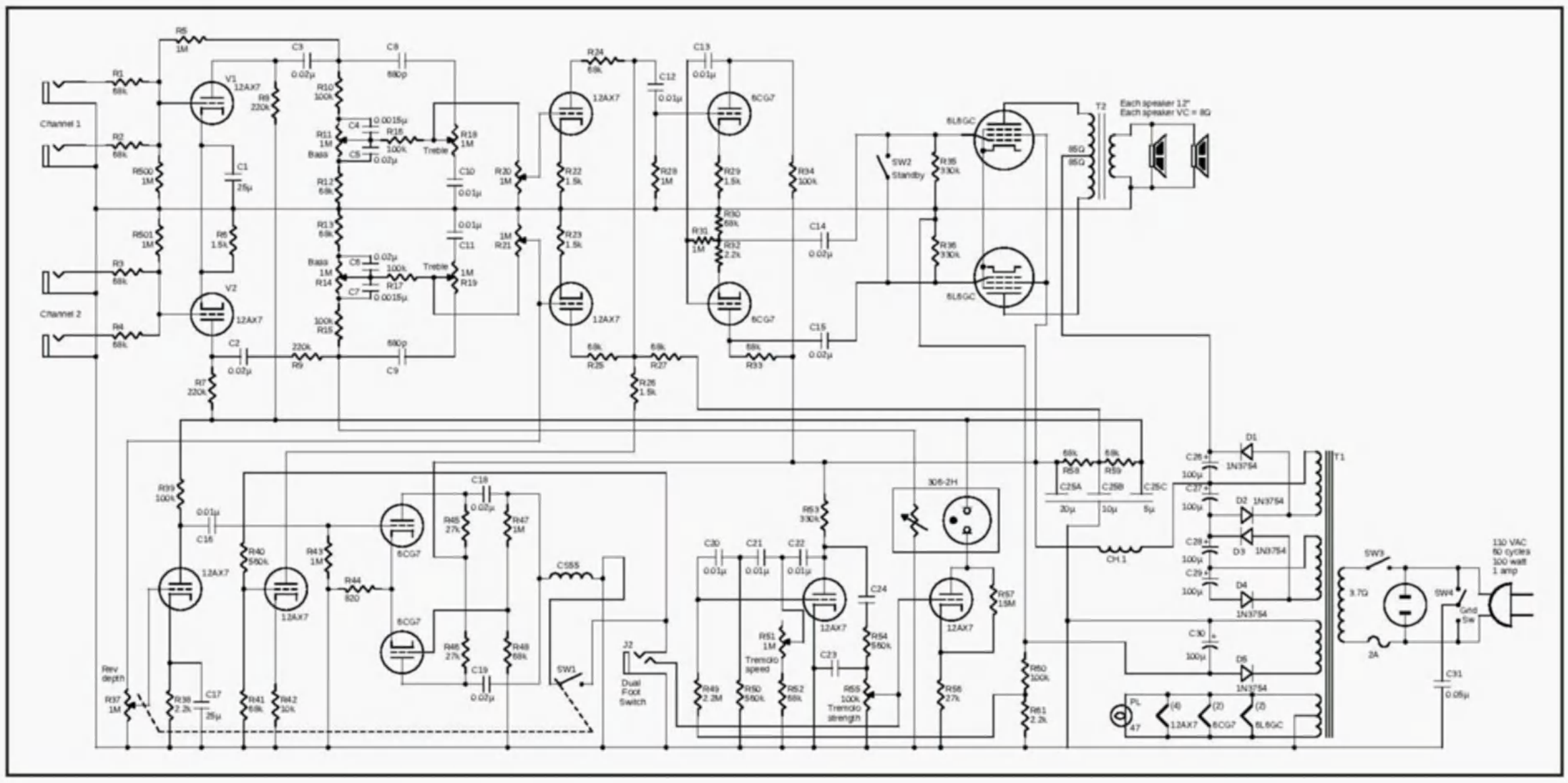


Figure 2: This Silvertone amp uses a voltage-doubler power supply.



Photo 3: The four large tubular electrolytic filter capacitors are part of the voltage-doubler.

Since in normal operation, these amps had 460 V on the plates of the output tubes, a 1.5 kV rectifier would give a bit over a 3:1 safety margin. Replacing the shorted diodes with the 1 kV units upped that safety margin to over 6:1. (By the way, the first Twins used hollow-state rectifiers, and later models used either the full-wave circuit shown in Figure 3 or a bridge rectifier). When a bridge rectifier was used, you could replace each diode with two 1 kV units in series.

Another problem that sometimes showed up in Twins was a shorted bias diode. This sometimes resulted from old output tubes shorting internally, or from a short in the electrolytic capacitor that served as a bias supply filter. In either case, the shorted bias diode (shown in the bottom right of the Figure 3 schematic on page 63, above the power transformer) let the output tubes draw too much current, and it was a race to see whether the output tube or the power fuse would fail first. Whenever I had to replace a bias diode, I always replaced the bias filter capacitor and the output tubes, since either of these could have become intermittent, possibly endangering the new diode.

Sometimes Twins would be brought to me with a complaint of a frying noise. This would go away if the Reverb control was turned down. It was caused by corroded RCA jacks on either the amp chassis or on the reverb tank. Or it could be caused by corroded RCA plugs on the reverb cable. In either case, rotating the plugs while leaving them plugged in would provide a temporary fix. Spray cleaner

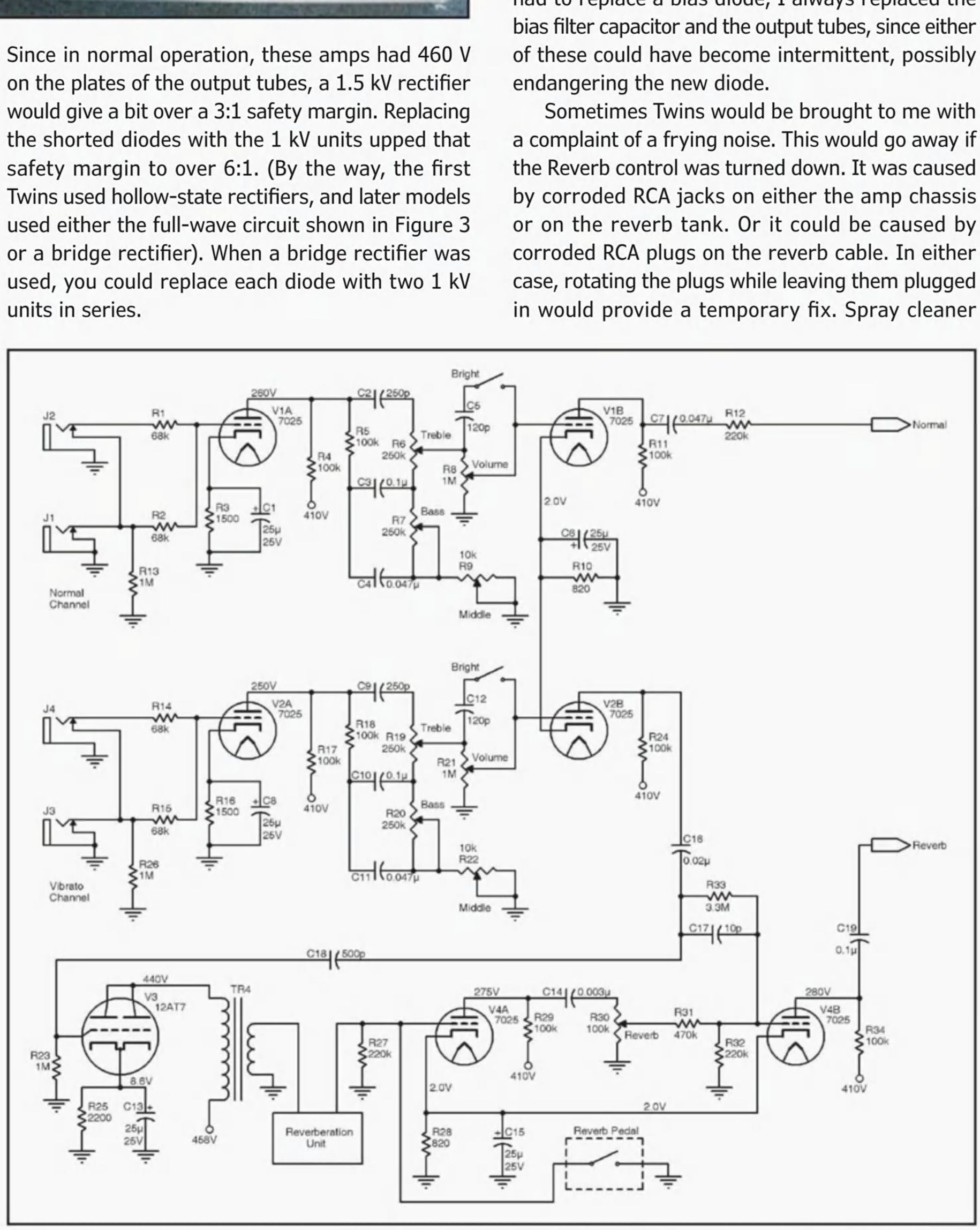


Figure 3: The Fender Twin Reverb amp has long been a best-seller. (the schematic coninues on page 63)

applied to the jacks, followed by plugging and unplugging the jacks a dozen times or so, was a longer-lasting solution.

But the best fix was to replace the jacks or plugs. I should add that when I asked the amp owner, I usually found out that the amp had been stored in a basement or a garage. The moisture in this environment (or in any location near the ocean) promotes corrosion in any contacts: It's not just a Fender problem.

A friend once brought me a Fender Vibrasonic, an amp very similar to the Twin, which made an intermittent crackling noise. I found a slight indication of corroded reverb connections, but not enough to explain the degree of noisiness he reported. Checking further, I found that three of the four output tubes were weak, and the fourth sparked internally when I tapped it. The three 12AX7s in the reverb channel were all microphonic.

This was an unusual case, in my experience, as usually when I have needed to replace tubes, either a single (or rarely, two) 12AX7s were microphonic or an output tube was very weak. Five bad tubes in one amp at the same time has not been common.

Engineering Issues

A few common amp problems have been traceable to poor engineering or quality control. Shortly after Magnavox bought Ampeg in 1971, it started sourcing film capacitors from the companies that supplied the Magnavox TV factories. These capacitors could be recognized by their rectangular shape and banana-yellow color. They were not as reliable as the caps Ampeg had formerly used. Also, the VT22 "distortion" guitar amp and several similar models used a shield attached to the chassis that was meant to be "un-telescoped" around the 12DW7 channel 1 input preamp tube. If the shield was not extended, the amp could oscillate, usually above the audible range, causing a particularly nasty

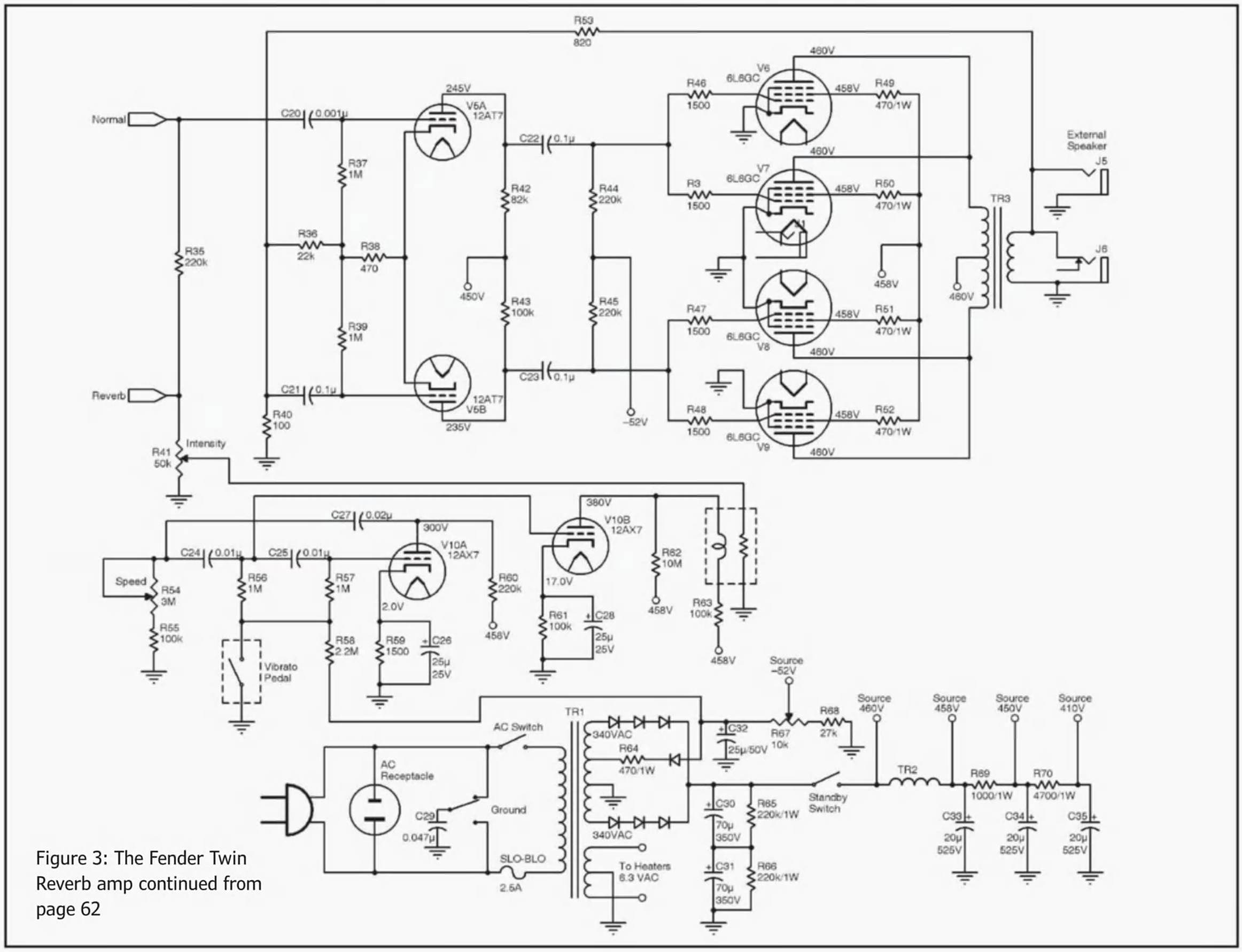
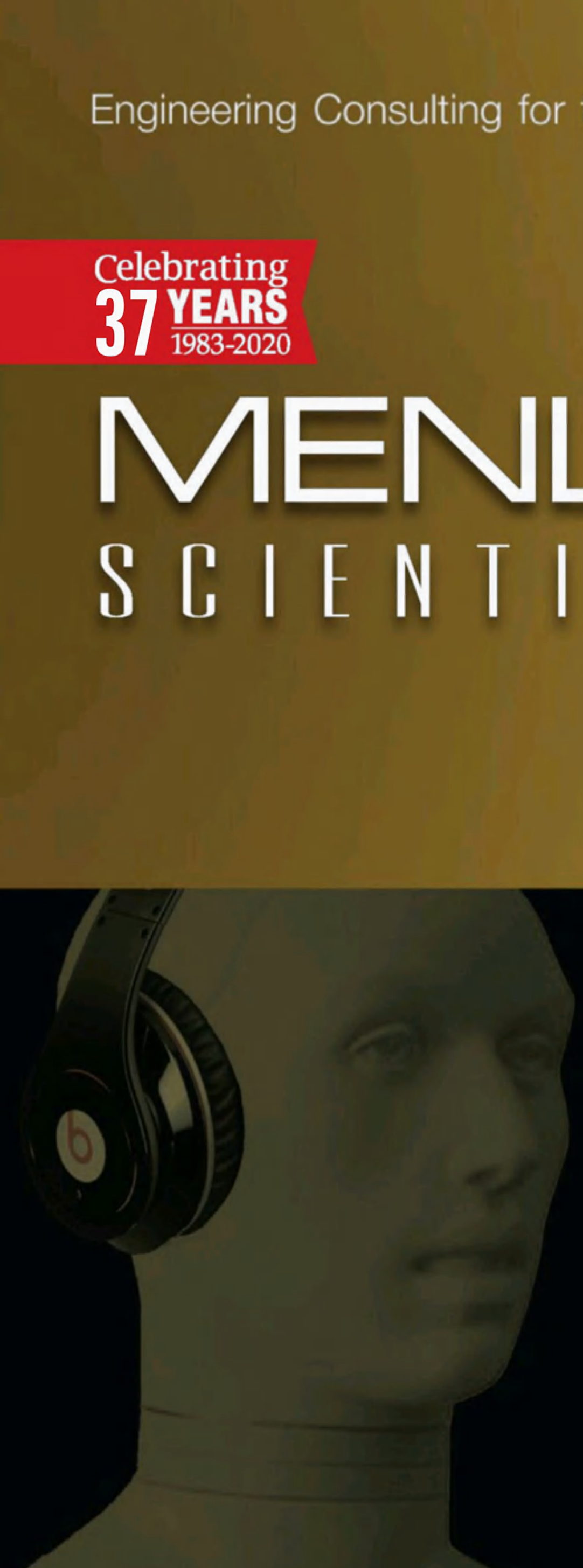


Figure 3: The Fender Twin Reverb amp continued from page 62

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distortion. Ampeg released service notes about this, and suggesting replacement of two resistors and two capacitors in the event that oscillation remained.

From time to time, a customer would bring in an amp that had a specific horrible smell that could be traced to charred output-tube sockets. Almost always, the customer had replaced a 6L6GC, 6550A, or KT66 output tube (7AC base) with a 7027 output tube that was used in many Ampegs (8HY base) or an EL34 (8ET base).

As you can see from **Figure 4**, the different base connections are not interchangeable. This is especially true for amps in which the designers have used the "open"connections of pins 1 and 6 of a 6L6GC as tie points for other circuitry. When tube sockets have been changed into crispy critters in this way, there is no good solution except to replace the sockets—not a quick or easy job!

In the early 1970s, several amps began to incorporate channel switching. This was a

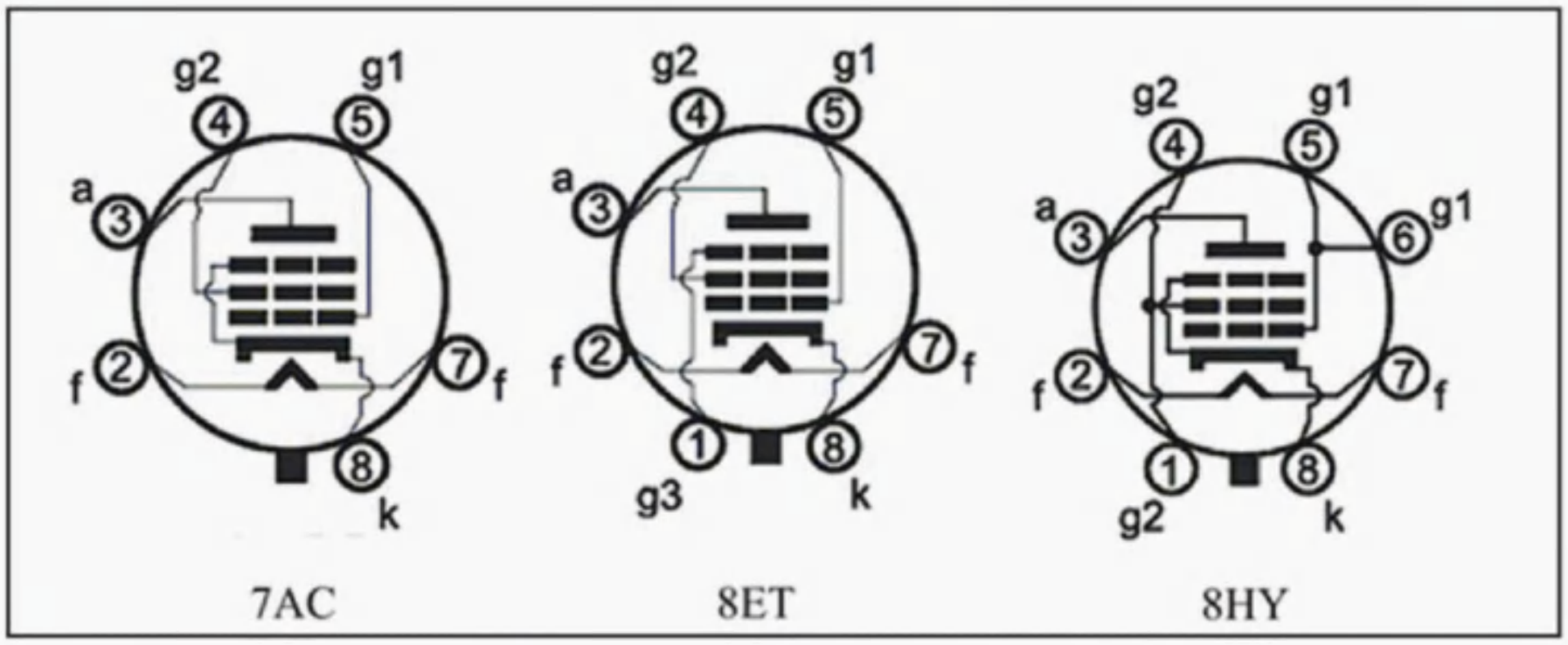


Figure 4: The base connection diagrams of the the 6L6GC, the 7027A, and the EL34 do not match each other.



development using a footswitch to activate a mini-relay in the amp chassis to switch between the "clean" or "low-gain" channel and the "distortion" or "high-gain" channel. Relay failure was a common problem in early amps of this type. Since reed relays were used, the contacts were in a sealed glass capsule, which prevented corrosion. But sometimes the relay coil opened or the footswitch contacts failed to operate. In most cases, the symptom was no signal for either the "lead" ("solo") or the "rhythm" position. The solution was to replace the relay.

General Problems

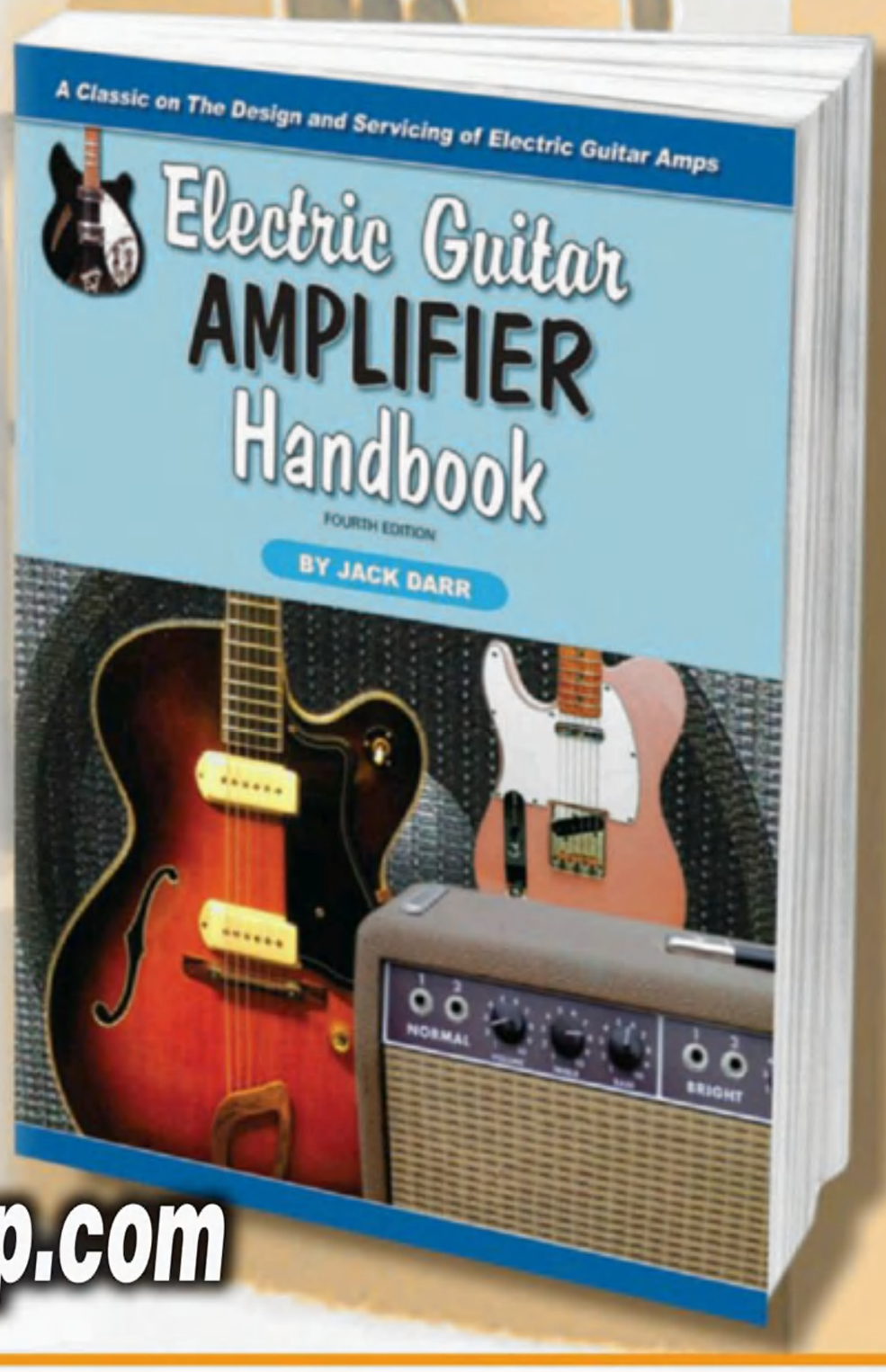
A few problems were common to guitar amps in general—not necessarily to a specific brand or model. Especially lower-priced amps, or any amps stored under the wrong conditions, could develop noisy contacts on the preamp tube sockets. This manifested as a "static-ey" noise, and could be pinpointed by wiggling the preamp tubes. If the wiggling caused a frying noise, the socket needed cleaning or replacing. Cleaning could be done by spraying contact cleaner into the socket holes, then inserting the tube and pulling it in and out repeatedly. If you did this with the power on, you could hear when the contact was clean, as the noise would stop. In really bad cases, the tube socket would have to be replaced.

Noisy contacts in volume and tone controls (or reverb and tremolo intensity controls) can be a problem in any amp. Cleaning the controls can fix this. Sometimes just operating the control rapidly to and fro is enough, but sometimes spray cleaner is needed. In this case, you have to open the amp chassis so you can see the back of the controls. If the control casing has an opening adjacent to the terminals, spray the cleaner in it, then operate the control back and forth. If there is no opening, you have to replace the control. In no case can you fix a noisy control by spraying cleaner near the shaft on the front of the amp. At best, nothing will happen, but at worst, you can wash out some of the lubricant around the shaft.

I'm sure readers who have experience in hollow-state guitar amp repair have other experiences that they could share, but maybe the ones I've mentioned will be helpful to anyone who fixes misbehaving hollow-state guitar amps.

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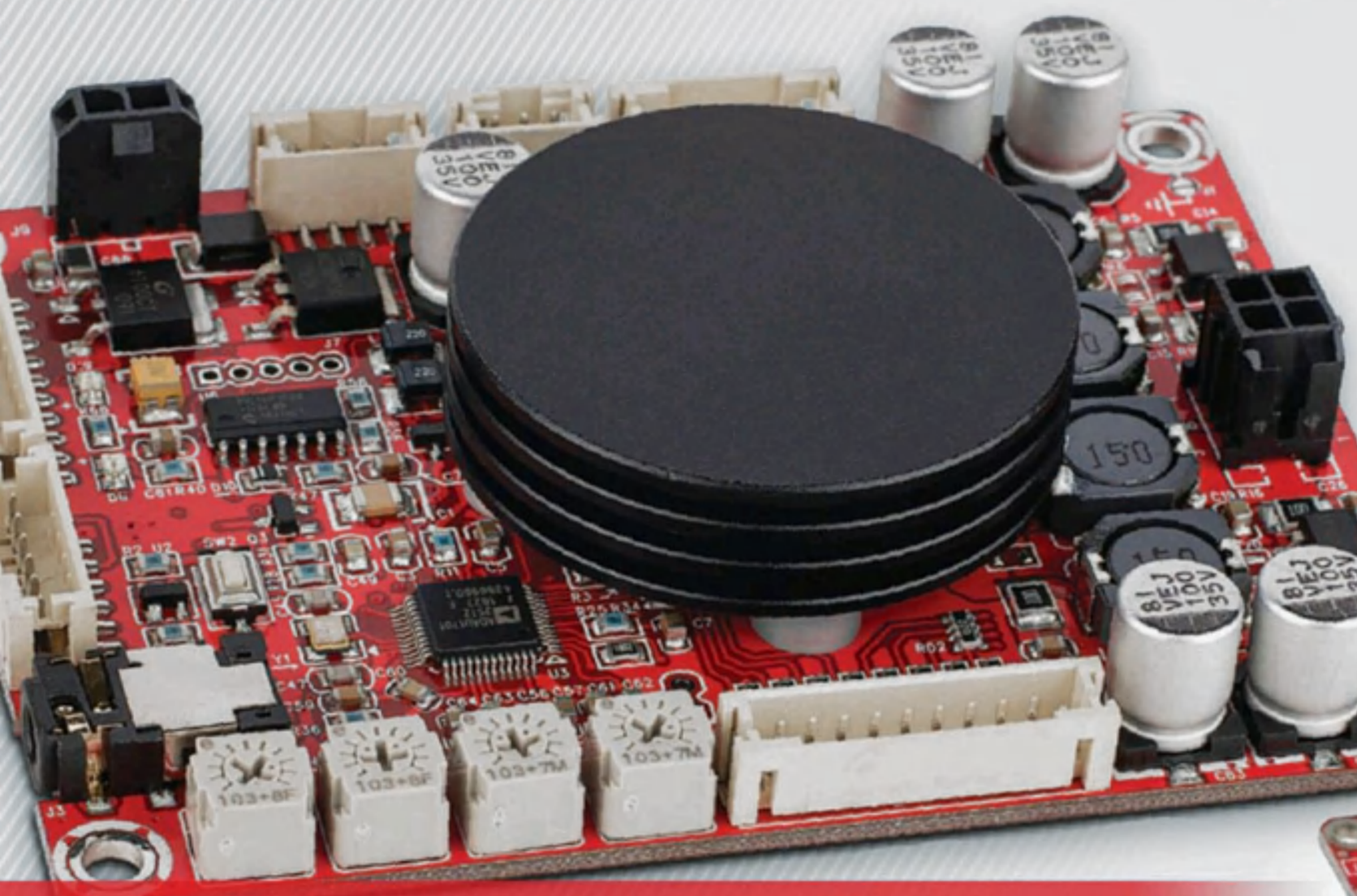
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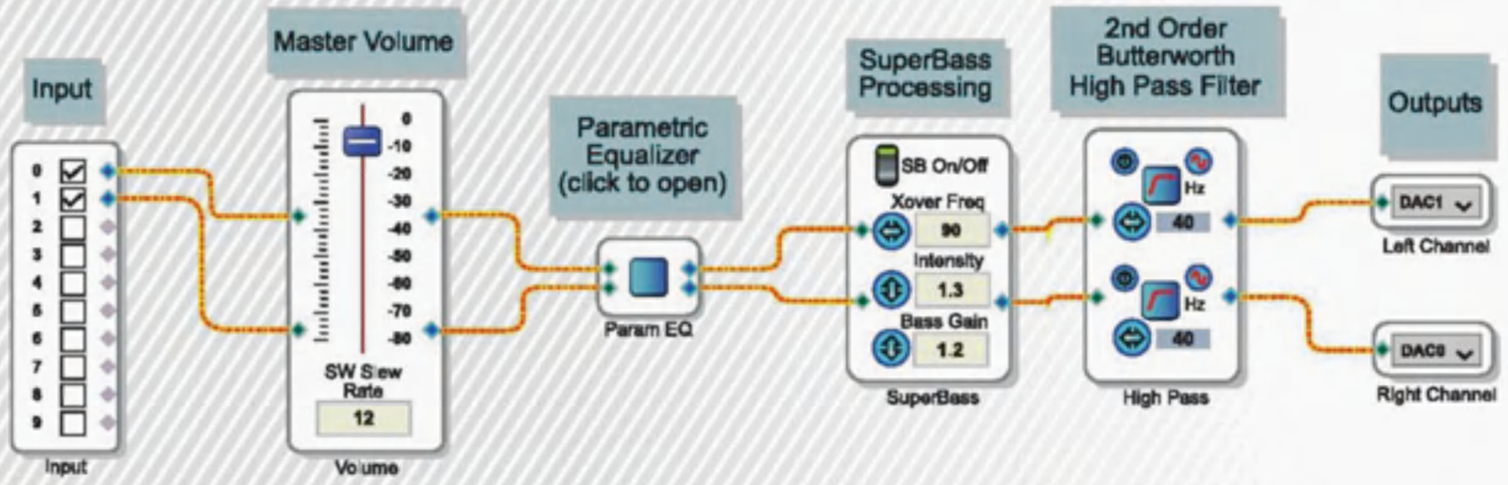


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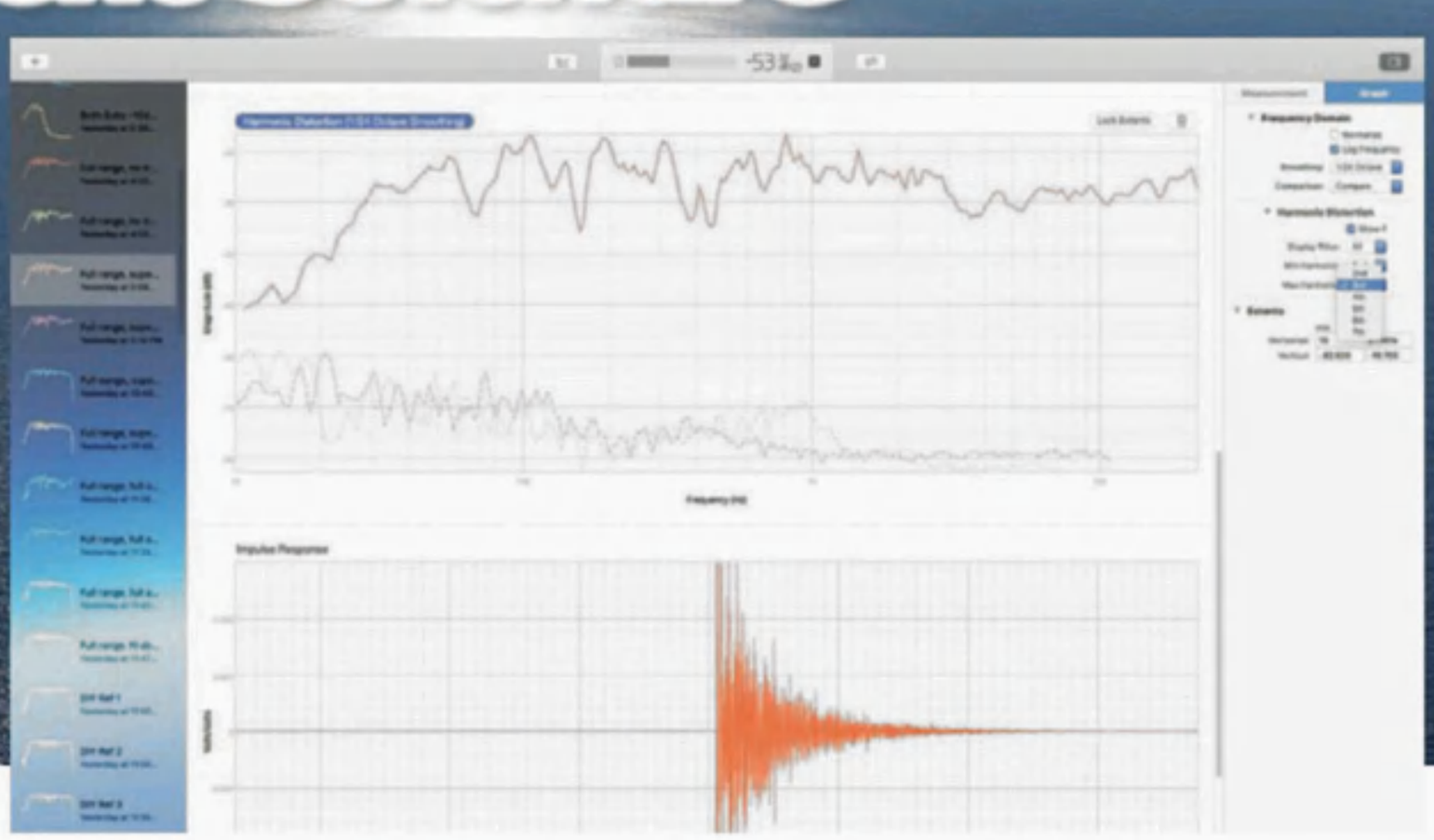


Photo 1: The simplistic FuzzMeasure hosts beautiful graphs in a straight-forward interface, yet still has heaps of advanced measurement options.

By
Luke McCready
(United States)

FuzzMeasure acoustic measurement software stands out for its high-quality graphs and for tucking an immense amount of control into a sleek, minimal interface. But just how useful is FuzzMeasure to the studio owner or audio enthusiast? Luke McCready ran the software through its paces to find out.

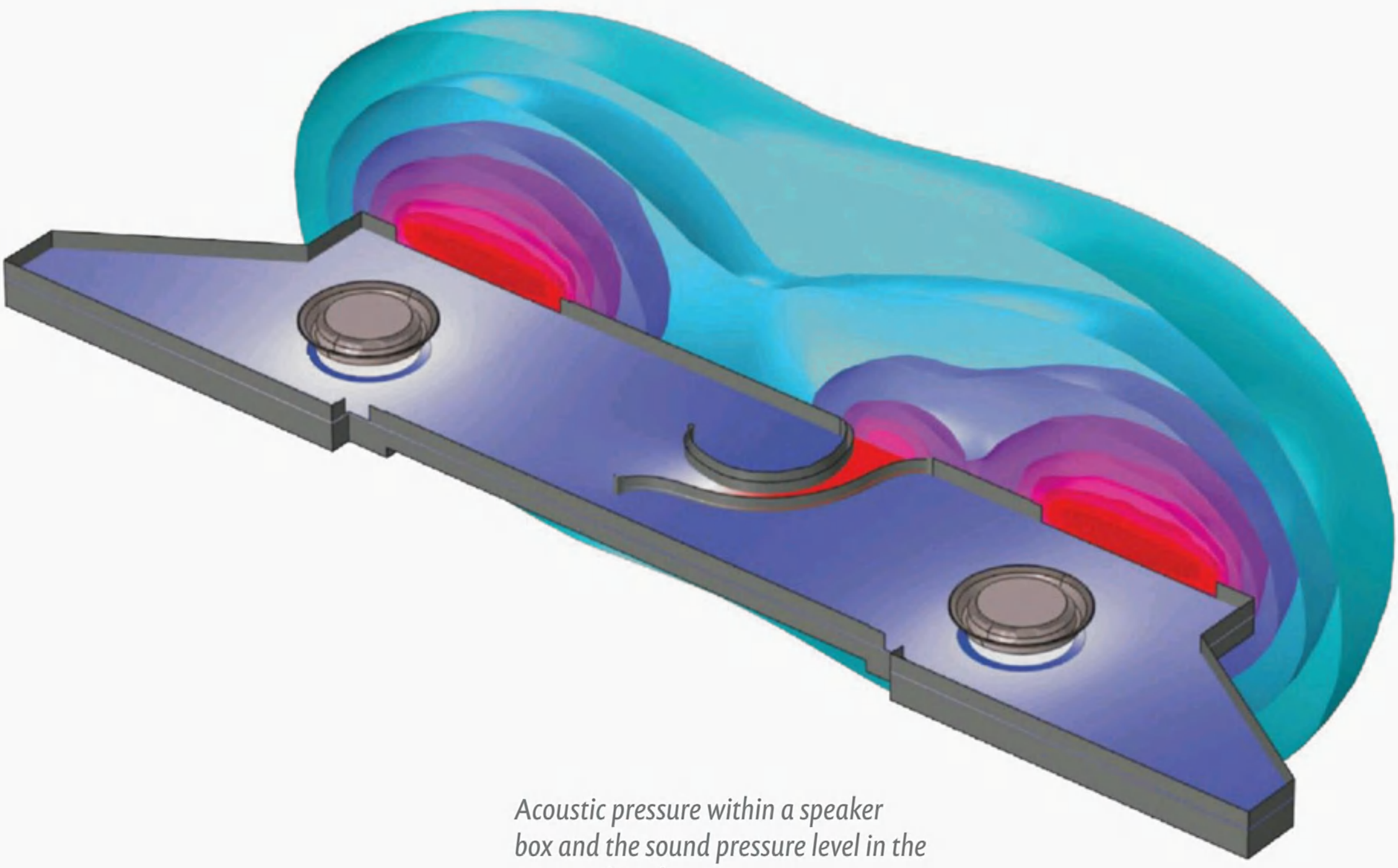
Studio setup is hard. Really hard. Every speaker has a flawed frequency response, every room has acoustic problems, and speaker and especially subwoofer placement are not intuitive but only mildly predictable even to studio design experts. But for a mix or mastering engineer to deliver accurate audio that translates to a variety of playback systems, guessing isn't good enough. One needs to know what problems exist in the room, and whether any new adjustments are helping or hurting.

FuzzMeasure, an audio analysis program, gives us the tools to know what is happening in a room. The concept is simple: Use the application to play test tones through the speakers, record back those sounds with a measurement microphone, and the software interprets the microphone's results to graph out frequency response aberrations, impulse response data, harmonic distortion, and much more. With measurements, one can place speakers and subwoofers with precision, accurately set crossover settings, and even dial in equalization using additional hardware or software to achieve a flatter, more accurate sound.

From Røde Microphones to RøDeTest

FuzzMeasure is a product of the new RøDeTest division of Røde Microphones, a subsidiary of Freedman Electronics, which is also the owner of Aphex, Event Electronics, and SoundField. Peter Freedman started Røde Microphones in 1990 to design and manufacture mics in Australia, aiming for low-cost, high-quality products with home recording enthusiasts in mind. Freedman began experimenting with microphones for cameras in the early 2000s to better record his son's sporting events. This exploration lead to a broad product line of successful on-camera microphones, extending Røde's reach outside the studio. Camera mics range from $60 to $600; studio mics, including tube and ribbon mics, range from $150 to $800; and Røde's

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most expensive mics at $1,000 each include a 22" shotgun condenser and a four-channel condenser for Ambisonic recordings targeting artificial reality applications.

But the story of FuzzMeasure doesn't start with Røde. Canadian software designer Chris Liscio created a program to help test the impedance of a DIY speaker project he was building. When Liscio realized there could be a market for measurement software, he spent months developing the program for its initial release in 2004 as FuzzMeasure. The market showed interest in more than just impedance, and Liscio broadened FuzzMeasure to include more functionality, growing into the audio and acoustics measurement software it is today.

Liscio, behind the brand SuperMegaUltraGroovy, began blogging about his progress in 2003. Early posts not only include insights on the challenges and developments related to FuzzMeasure, but his personal computer woes, a baby announcement, and even recounting dates with his wife. But the blog shifted over time to include his company's other projects, starting with TapeDeck, released in 2008. And in 2009, Liscio debuted Capo, an award-winning program designed to help amateur musicians learn to play instruments by ear.

Liscio shared on his blog that he thought Capo's success would enable him to hire development help so he could return to FuzzMeasure. He wrote that he missed the old days of losing himself in development for a week or three, making rapid progress on new software features, but that just wasn't the reality running a larger software business. In 2014, Liscio handed off the administration of SuperMegaUltraGroovy to return his focus to product development and, in 2018, Røde acquired FuzzMeasure from SuperMegaUltraGroovy.

The strategy, confirmed by Peter Freedman, was to use FuzzMeasure as the foundation move for RødeTest, a new measurement division, adding a range of professional acoustic measurement microphones and accessories in the near future.

FuzzMeasure 4.0, the program's most recent major release, debuted in 2015 with numerous UI updates, more usable graphing functions, and many quality-of-life changes. It's unclear how much of FuzzMeasure's current functionality, if any, Røde has added. However, the accompanying documentation is significantly more helpful than it was in 2015, and RødeTest muddies the waters with its Software Maintenance Subscription, though FuzzMeasure is a perpetual license.

Røde has already rolled out additional hardware to aid in acoustic measuring, including a Røde single-input audio interface. Measurement microphones have been in development at least since 2018, though a product has yet to surface. Fortunately, FuzzMeasure is compatible with all varieties of measurement mics and can load in their unique calibration files.



Photo 2: I measured my monitor surrounded by absorption to see what the low-frequency-filtering input on my amp is doing.

A Competitive Field

FuzzMeasure is far from the only acoustic measuring software available. Despite its complexity, Room EQ Wizard (REW) remains ever-popular because it's available on Windows, macOS, and Linux, and it's free. AudioVero Acourate is a mid-priced option that features an interface and graphs as dated-looking as REW. Smaart by Rational Acoustics is a comprehensive tool used in the live sound environment, specializing in large public address systems including concert line-arrays. What separates the macOS-only FuzzMeasure is that it's relatively inexpensive at $100 for consumers and $500 for commercial use, its graphing looks the way you'd expect in 2020, and it's able to provide deep complexity from within a simple and easy-to-understand interface.