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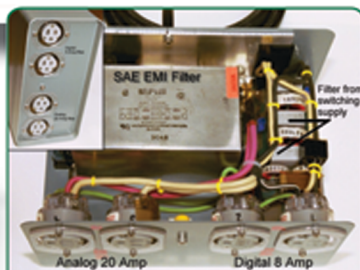


## Modern Makeover for A Dynaco Classic

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# Vintage Dynaco SCA 80

By Bruce Brown

This Dynaco devotee shows you how to put a classic amp back into good use.

Most *audioXpress* readers are familiar with the products manufactured by Dynaco from the '50s to the early '70s. During this time Dynaco sold thousands of factory-built audio products and kits. These varied greatly, from tube amplifiers such as the Stereo 35 and 70, to integrated tube and transistor amps, to the monster ST 400 and 416. Dynaco offered real value and fun for the audio hobbyist.

There are probably several hundred modifications and updates for Dynaco equipment, with the largest majority for tube equipment. Online auctions for used Dynaco equipment often offer less popular items at a very reasonable price.

*audioXpress* has featured many articles on updating and rebuilding these older products. The very nice article written by Daniel Dufresne on the FM-5 and AF-6 tuners in the August 2003 issue inspired me to write this article on an often-overlooked integrated transistor amp, the SCA 80 (Photo 1). This amp combined the PAT 4 preamp (Dynaco's first transistor preamp) with the ST 80 transistor power amp.

the specifications of the SCA 80, you might be surprised to learn that it provides 40W RMS per channel into an 8Ω load with a total harmonic distortion of 0.5% from 20Hz to 20kHz.

It features much flexibility, with inputs for magnetic phono, tuner, tape, spare aux, and tape monitor. It also has a "special" input that you can configure for either a tape head input or a second phono input. In addition to the volume, balance, treble, and bass controls, it also features front panel tape monitor, loudness compensation, rumble filter, mode switch, and either a main/remote speaker connection or four-speaker "surround" system, depending on the model. I prefer the sound of the 80 se-

ries amps to the ST 120 amps. There were probably ten times as many ST 120s built as there were ST 80s, but even though the circuitry is similar, I think the 80 has far better bass response and is less "tiring" to listen to for long periods of time (my opinion, not based on any specs).

The preamp section of this amp uses two transistors in the low-level inputs and two others in the control circuit. These use a DC feedback loop to stabilize operating conditions, as well as an AC feedback loop to provide optimum audio performance. The tone controls provide a wide range of adjustment as well as a center "flat" setting. The amplifier section provides protection while delivering exceptional performance.

Transistors Q1 and Q2, which are a direct-coupled feedback pair, provide stability and linearity and drive Q3 through Q6. The amplifier circuit



PHOTO 1: Dynaco SCA 80 amp.

## FEATURES

Audiophiles often ignore this unit, but it represents a compact, relatively powerful integrated amp—ideal for an office, bedroom, or dorm room. If you look at

is designed to reduce the current through the output if drive signals are excessive or under heavy loads. The output transistors are operated without quiescent current, which limits heat rise and eliminates the need for temperature compensation devices. The output capacitors, C7s, block DC from the speakers, yet allow unrestricted low-frequency response.

## BACKGROUND

The first model (relatively rare) was the SCA 80, which had main and remote speaker outputs. The later, much more common, SCA80Q changed the speaker switching to include the DynaQuad circuit, and allow two rear speakers for an early "surround" sound. As an interesting sidelight, the Q model was rated at only 30W per channel, even though the circuitry was essentially identical. This was in spite of the Q model having heavier duty transistors for the driver and outputs.

I have repaired both and can tell you that either of these models actually will put out 35-40W RMS into an 8Ω

load. When restored, these amps will provide very pleasing sound and, combined with some vintage speakers, offer flexibility (not to mention the fun of resurrecting something that might have ended up in a landfill). With a little work and small investment, you will have an amp that will provide many more years of faithful service.

*Safety note:* You will be working on a live powered chassis with large storage capacitors. Use the same cautions you would working with high voltage circuits.

You will first need to locate an SCA 80 or SCA 80Q and the correct factory assembly manual. Many are listed in online auctions (often with one channel not functioning) and can be obtained inexpensively. As long as the power transformer is good, they are candidates for this restoration.

In addition to the two factory-assembled models, there are also kit versions. Usually factory-built units have a sticker on the back panel. This may

be missing due to the age of these amps, but you can usually tell by examining the wiring: kit-built units used mostly red wire for one channel and green for the other.

Factory-built ones used multicolored wiring that is identical for each module (*Photo 2*). You may also find that a previous owner "updated" the amp with various types of "audio-ophile" wire. It can be a rat's nest! I have found many times that modified amps are not wired correctly, or virtually all the soldered joints were suspect. If this is the case you should consider a complete rebuild of the chassis or find another one to modify. Make sure you ask questions if an auction listing does not specify conditions.

## ASSESSMENT

Once you have your unit, you will need to pull the cover off and evaluate the operating condition. You should have a DVM (digital voltmeter), and, ideally, a signal tracer, as well as a set of inexpensive test speakers and a signal source such as a tuner or CD player. Check to see that the proper 2A slow-

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blow fuses have not been replaced with a larger one.

You will also need some contact cleaner. (I use a product named De-toxit 5.) Clean all the pots and switches thoroughly, then connect your speakers and audio source. If you have a Variac®, bring up the voltage to about 40V and then to 60. This will allow the power supply to charge and reform the capacitors.

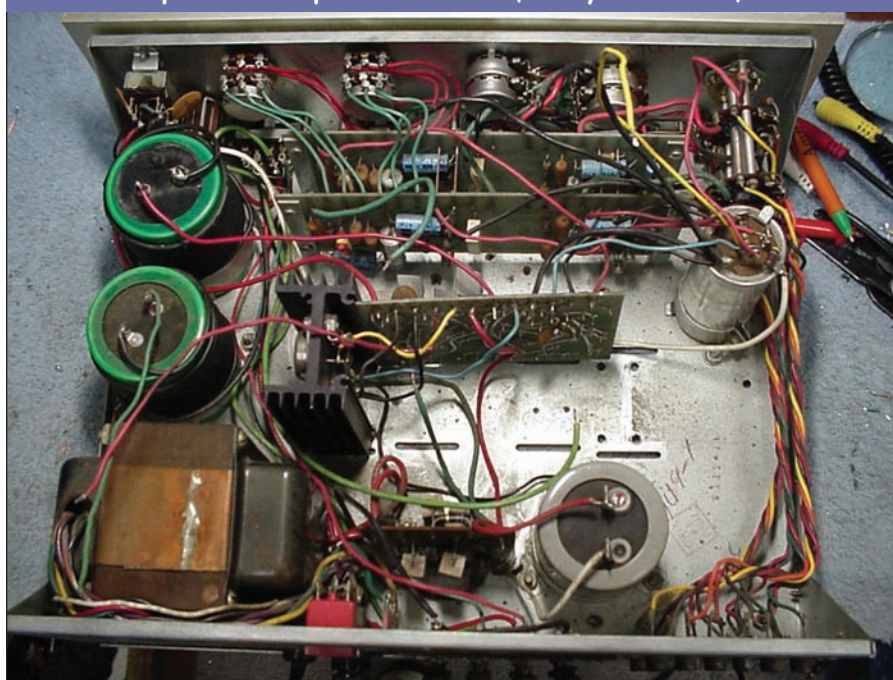
If you do not have a Variac, turn the power on, watch the components, and listen for bad sounds. If you experience excessive hum from either of the speakers, you probably have a shorted output transistor. Isolate the channel, power the amp down, and remove the power lead from the large single filter capacitor (doing this isolates the power transistors). This power lead comes from the top-most power transistor in each channel and connects to the positive lead of a large single filter capacitor.

Re-power the amp and continue with your testing. If you do not have smoke at this point, it is time to check some voltages. This may sound a little bit cavalier, but the protection circuitry built into this amp will generally prevent damage, and if the fuse holds, at least the power supply is functioning correctly. If you have a Variac, bring the AC up to 110V.

To evaluate the state of your amp, connect your DVM or voltmeter ground to the chassis and measure voltage with the positive lead, as follows:

1. Connect to the positive lead of the large filter capacitor C9 (the one sitting by itself). You should have approximately 70-75V DC. If so, the main power supply is functioning.
2. Connect to the positive post, one at a time, to each of the other large filter capacitors. You should have between 34-36V DC at each one. If you do not have these approximate voltages, you have a defective power amp channel. I will discuss this repair later. If you previously removed power from the power transistor, you will not have this voltage.
3. Next go to the smaller can capacitor. At the lead connected to the triangle marked section you should have around 70V DC. The one marked with the square should be around 24V and the one marked with the circle should be around 17.5V DC.
4. Now connect your positive lead to either of the power amp board's terminal 6, remembering the board has opposite orientation, and you should have about 37.5V DC at each of these points.

PHOTO 2: Amp with one amp module removed (factory-built version).



5. Next go to terminals 8, each of which should have about 36V. If either of these last two does not measure at 36, you have a bad transistor or a bad Zener diode on the power amp board. I will also discuss this fix later. (If you disconnect the power lead from one of the channels, only one of these two points will have the correct voltage.)

Proceed with audio sections testing. First set the input selector to the audio source you are using, then turn up the volume slightly and check the following:

1. Connect your signal tracer to ground and touch the probe alternately to each of the center terminals of the volume control. If you have signal here, the input switching system is working.
2. Now alternatively connect the probe to terminal 12 of each of the amp boards (PC-18). If you have good clean signal at this point, the preamp board (PC-17) is operating correctly. I have never had one of these amps with a preamp board failure, but if you have distortion at terminal 12 of either of the amp boards, you will probably need to troubleshoot the preamp (PC-17) and replace some small electrolyt-

ics or a preamp transistor. You may also choose to upgrade the capacitors on this board (see Update sidebar).

## BOARD UPGRADE

The issue with most SCA 80s is the PC-18 amp board, and Dynaco made several versions. The early version used dark brown phenolic circuit boards with metal driver transistors that had "top hat" heatsinks (Q3 and Q4). The next version was a light tan board, again with metal driver transistors.

The final version of these boards was a translucent green fiberglass military-type board, which usually had plastic cased TO220 driver transistors with bolt-on heatsinks. This is the most stable version of this amp module, and I will update the older boards to the later version. If you have the later version, follow these troubleshooting details and replace defective and outdated components with the parts list.

You will need to remove each of the amp modules from the chassis for the following tests and modifications:

1. From PC-18, unsolder the wires that go to terminals 1, 6, 7, 9, 10, 11, and 12.
2. Remove the four bolts that hold the heatsinks in place from the bottom

of the chassis.

3. Unsolder the two remaining wire connections from terminals on PC-19 (the power supply board). There is also a ground wire that connects the lug on the base of C-19. The amp module is now free from the chassis. It is time to troubleshoot and update this module.

Driver transistors, especially Q3, are typically a problem on these boards. If any of your transistors is bad, you will also need to replace the Zener D1. I have included these in the parts list and I recommend that you replace them during the rebuild regardless. Early units used 2N3055 output transistors, while later ones used 2N3772s. The parts list calls for 2N3772s, which I recommend you use to replace all the outputs, regardless of the condition of the existing outputs in your amp.

Using your ohmmeter or a transistor tester, check each transistor while it is still in the circuit. If your amp was not working correctly, you will usually have one or two bad drivers and one or both outputs blown. You should also check the silicon diodes located at D2 and D3. In addition to the transistors, also replace some of the troublesome and outdated electrolytics on these boards.

Remove capacitors C3, C4, C5, and C6 from the boards, as well as the driver transistors Q3 and Q4, and the Zener diode D1. You also need to remove the 27pF ceramic capacitor C13, but save this for reuse on the back of the board. It is a good idea at this point to turn the board over and use a round toothpick and your iron to clean out the holes where all these components were mounted.

1. Replace C3 and C4 with new 33 $\mu$ F 50-100V non-polarized electrolytics.
2. Replace C5 with a new 470 $\mu$ F 16V electrolytic, making sure to observe correct polarity.
3. Where you remove C6, replace it with a 0.47 $\mu$ F 100V Mylar capacitor. On some boards this was a non-polarized cap, on some later ones, a Mylar was standard. There is no polarity on this one so install either

PHOTO 3: Soldering the pins on the board underside.



way.

4. Install the 68pF ceramic where C13, the 27pF, was located.
5. Install the new 5.1V Zener at D1 (be sure to observe polarity).
6. You will need to bend the leads of the TIP 31 and 32 to correctly align the pins for mounting in Q3 and Q4. Be sure to get the base, collector, and emitter leads in the correct holes. Q3 is the TIP 31C and Q4 is the TIP32C. You should also attach your new heatsinks to these transistors before you mount and solder them in place. Use a thermo pad on each of these, although it is not necessary to use insulating hardware.
7. Turn the board over and solder the 27pF ceramic between pins (*Photo 3*) to stabilize oscillation that may occur with some input sources.

Replace each of the output transistors mounted on the attached heatsink, using new thermo pads instead of the original mica insulators. Refer to the pictorial in the manual to get everything back in the right place. It is also

a good time to check the 0.47Ω 1W resistors on the back of the heatsinks. Many times these will be open or burned if the amp had a catastrophic failure.

## TESTING

You now have one of the amp modules completed (*Photo 4*). Complete the same process with the other module. Then, carefully examine each module for solder bridges and make sure the new heatsinks do not contact any of the component leads.

Measure the resistance of each of the board-mounted resistors and replace any that are out of tolerance. Usually there are not many problems with these, but a little prevention here can save you time later. Before reinstalling the modules, it is a good idea to go through the amplifier and resolder all the attachment leads on the other circuit boards and controls.

Before connecting the leads to the large output capacitors, it is necessary to test these units. If either is faulty, the amp will not function correctly and your voltages may be considerably off.

I re-form both of these with a power supply capable of providing 75V. Connect the positive and negative leads of the power supply to the positive and negative terminals of the capacitor, slowly bring up the voltage over several minutes, and note the charging current. No current draw indicates an open cap; excessive current (greater than 25mA) indicates a shorted or leaky capacitor.

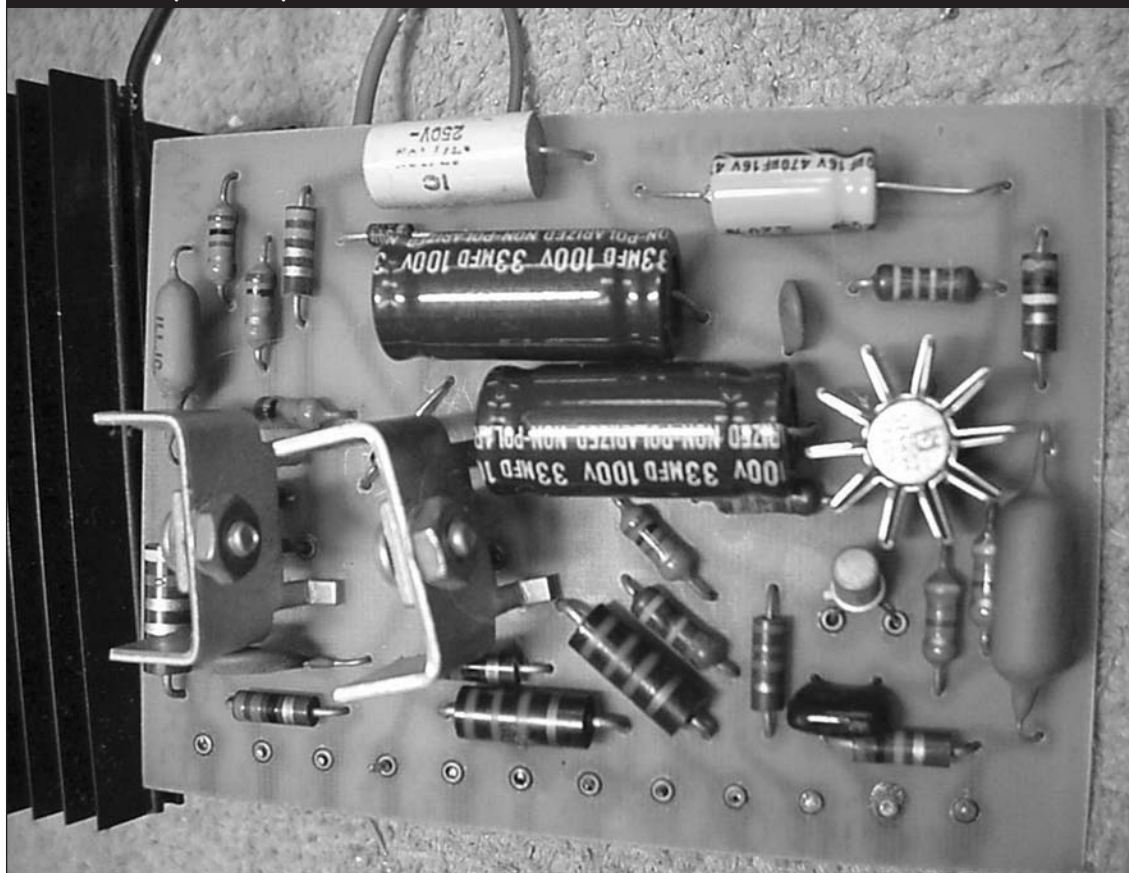
If you cannot keep the current below 25mA, you must replace the capacitor. An exact duplicate is very hard to find. Or, you can charge it with a 12V battery charger and then disconnect it and connect your DVM leads to the capacitor and monitor its slow discharge. If it takes a charge and slowly discharges, you can assume it will function correctly in the finished amp. If one of them is bad, you need a replacement.

New computer-grade caps of the same value will be smaller in diameter and will be difficult to transfer the large wire wrapped around it. This is an air core choke to stabilize the circuit and limit RF frequency flow to the outputs. You

can use the capacitor C9 from the power supply and transfer the wire to this, and replace C9 with a new 5000-6000μF 75-100V computer grade cap. You will also need the appropriate size clamp and drill new holes to mount it. Wire the new exactly like the original.

Remount and connect the leads to the modules and the power supply board. The most difficult part here is connecting the ground wire to the solder lug on the chassis attached to the power supply board. I usually

PHOTO 4: Completed amp module board.



solder these ground leads to the lug before I re-mount the modules on the chassis.

After you have completed all the wiring, check everything very carefully, using *Fig. 1*. A misconnected module can very easily destroy several of the semiconductors and require starting all over.

Reconnect your speakers and audio source and fire it up. A Variac is handy, but not necessary. You should be rewarded with very clean sound. It is also a good time to check the function of all the controls and switches. You may find the controls need a second shot of contact cleaner to ensure quiet operation.

This amplifier will drive most reasonably efficient speaker systems. I like the sound of Dynaco A-25s or vintage Advents. (These are also reasonably available and affordable in on-line auctions.) Consider purchasing a Dynaco FM-5 or AF-6 to match your newly refurbished amplifier. I recommend you perform the modifications covered in the 2003 article to complete your vintage system. In a sidebar (Dynaco FM5 Dial Light), I have included a further update for these tuners that makes their maintenance easier.

I hope you enjoyed this rebuild. I will be happy to try to assist you with any questions you might have. You can contact me at my e-mail address: Tuninfork@aol.com.

## DYNACO PREAMP BOARD UPDATE

Frank Van Alstine, of Audio by Van Alstine, was a leader for many years in the modification of Dynaco equipment. In the November 1986 (Volume 5, Number 11) issue of *Audio Basics*, he published an article about updating the PAT-4 preamp. You can go to his website ([www.AVAHIFI.com](http://www.AVAHIFI.com)) and search the back issues of *Audio Basics* to print out a copy of this article. Since the PAT 4 and the SCA 80 preamp boards are the same, you can use his recommendations to replace the outdated components, notably the small electrolytic capacitors in the preamp circuit, with tantalum caps.

He replaces some of the larger electrolytics with large film caps. The cost for the capacitors for this rebuild is about \$50. If you perform this modi-

## PARTS LIST

### Amp module rebuild (both modules)

4- 2N3772 power transistors (Electronic Goldmine, [www.goldmine-elec.com](http://www.goldmine-elec.com))

4- TO3 thermo pads (packs of 5 available from Mouser Electronics, [www.Mouser.com](http://www.Mouser.com), 526-NTETP0001)

2- TIP31C plastic TO220 transistors (Electronic Goldmine)

2- TPI32C plastic TO220 transistors (Electronic Goldmine)

2- 5.1V Zener diodes (Mouser 78-IN5231B)

4- 33µF 30-100V non-polarized electrolytic capacitors (Mouser 140-NPRL 50V33)

2- 470µF 16V electrolytics (Mouser 647-TVX1C471MAA)

2- 0.47µF 100V capacitors (Mouser 75-225P100V0.47)

2- 68pF 100V ceramic capacitor (Mouser 581-SR151A680J)

2- 0.47µ 2W resistors (Mouser 283-0.47) see text

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fication plus my modifications, you will have eliminated most potential problems with this vintage amp. This will much improve the sound of the preamp section of the PAT-4 and also the SCA 80.

Do not try to use his recommendations for power supply changes because they apply specifically to the PAT-4. He also describes how to rewire the input selector of the PAT-4 for an additional Aux input; this is also not applicable to the SCA 80 or Q.

## DYNACO FM 5 DIAL LIGHT

As many folks have discovered, the dial lights in the FM-5 and AF-6 are not readily available. The original factory lamps are significantly longer than currently available "fuse" type lamps. McIntosh does sell long lamps that will work, used in the early MX tuners, charging \$10 each.

A far less expensive solution is to use short "fuse" lamps with an adapter to lengthen them. My friend Larry uses two *fired* .22 caliber rifle shells, one slipped on each end of the lamp. If you use these cases, be sure to use *only*

fired cases, not ones with the bullet pulled and powder removed. Unfired cases will still have an active primer, which can be set off by the electric current of the lamp heat.

I prefer to replace the original holders with standard fuse holders and then use currently available "fuse" lamps. The parts you will need are listed at the end of this sidebar.

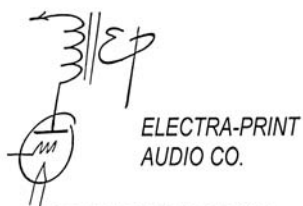
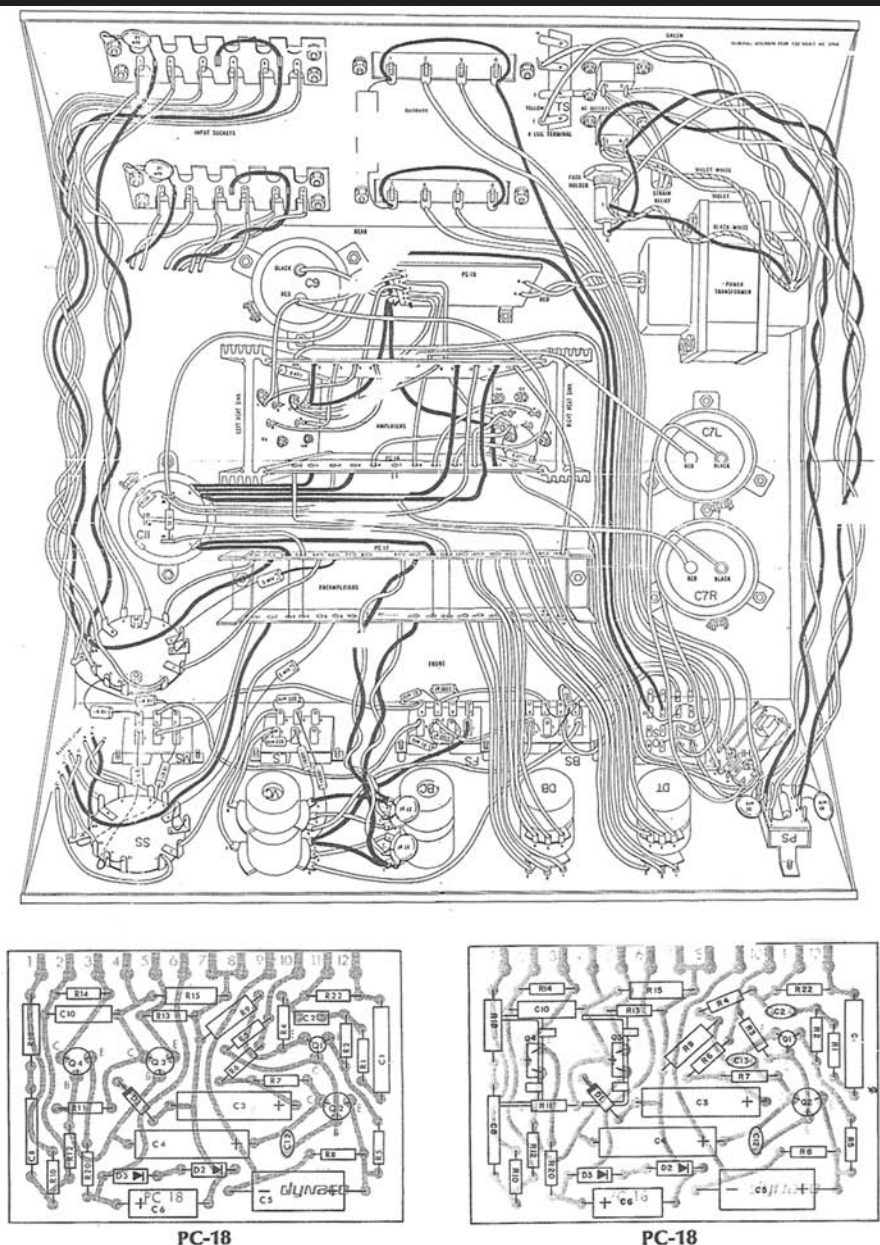
Another common issue is failure of the "tuned" lamp or "stereo" indicator lamp. Gregg Dunn's Dynaco site contains information on replacing these two grain-of-wheat lamps with LEDs. The tuner will not work properly with-

out them in place, so if your unit does not kick into stereo, replacing this lamp will generally fix it. I repeat those instructions.

For these procedures you will need the following:

- 2- 8V fuse lamps (All Electronics LPFS-8)
  - 2- fuse holders (All Electronics FHBL-3)
  - 2- Blue LEDs (All Electronics LED-94) (You can use white ones also)
  - 2- 15K ½W resistors (All Electronics) see text
  - 12" of ⅜" heat-shrink tubing.
- First remove the case cover, knobs,

FIGURE 1: SCA 80 diagram.



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and faceplate. Carefully remove the existing fuse lamps and holders. Using the same mounting hardware, install the new fuse holder. The ground lugs may need to have a longer piece of wire attached to them to work with the new holders. Once you have mounted them, install the new 8V lamps.

Now carefully remove the "tuned" lamp from its bezel. Cut the leads close to the bulb and attach the blue wire to the longest lead of the LED. Cut a piece of heat shrink and place on the wire lead before you solder the connection. Completely cover the exposed wire and LED lead with the heat shrink. Place a longer piece of heat shrink on the other lead.

Connect the resistor to this lead and the other end to the short lead of the LED. Completely cover the resistor and exposed LED lead with the heat shrink. Push the LED into the bezel. Follow the same procedure with the "stereo" light.

Now plug in and turn on the tuner and make sure that the LEDs function and the stereo light comes on when you tune in a stereo station (you should have the mode switch in the stereo position). If neither of these LEDs lights, or you do not have any blue leads on the grain-of-wheat lamps, you may need to reverse the leads. If neither works in either position, you have another problem with the tuner and it will require troubleshooting.

If both LEDs are on all the time, but become brighter when the tuner is tuned, you must increase the value of the resistors. This depends on the current draw of various LEDs. (I have used values as low as 10k and as high as 20k for them to light correctly.) Now clean the dial glass and reassemble the face and case. These simple modifications will make it easier to replace the dial lamps in the future, and virtually eliminates replacement of the tuning and stereo lights. *aX*

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**Acknowledgment**

I want to thank my wife Kristin for her understanding and wordsmithery of my articles. Without her help, you would not be able to decipher my ramblings.



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# Budget EMI Filter

By Rick MacDonald

Build this quality AC line filter for less than \$100 with surplus parts and recycled filters from old server switching power supplies. It will improve the sound of many systems and is easy to put together.

**N**ow that you have finished that new amplifier, done all that hard work, and spent way too much, you fire it up only to be disappointed because the sound is not what you had hoped. Or, after upgrading everything, your system just doesn't perform at the level expected. Well, you may be the victim of AC pollution, and not realize that it can degrade audio/video performance.

Our power grid is a huge electromagnetic network spewing out noise and interference across the spectrum. A good filter can control and reduce this interference, improving the sound (and picture) of your entire system. Of course, isolation transformers, line conditioners, and regulators are the best solutions for EMI problems and waveform regeneration, but at higher currents (15+A) they can be quite expensive. So if you need a way to clean up your AC, try this approach.

I constructed this mains filter (*Photo 1*) using an SAE two-stage EMI filter (bought surplus) and a line filter (*Photo 2*) pulled from a discarded server switching power supply. It has two analog outputs rated at 20A, and two digital outputs at 8A. A large ferrite core (#43) was looped through twice at the AC entry point, and came from the sample kit (*Photo 2*), also purchased from a local surplus.

If you have trouble finding these items surplus, Mouser and Digi-Key carry most of them. If you choose to build one, the circuit is simply a multi-stage pi filter (*Fig. 1*), and you can find

others on the net, but it's easier to buy them already potted in a metal case (*Photo 2*).

A computer recycler is a good place to start, as many servers use high current switch mode power supplies and rely on filters to keep switching noise from being imparted back to the AC line. That also makes them perfect for isolating

**PHOTO 1: Top view, front view (inset).**

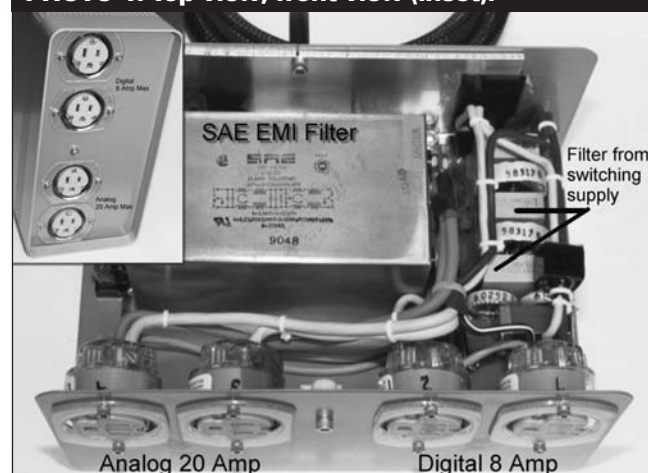


**PHOTO 2: Ferrite cores and filters for EMI.**

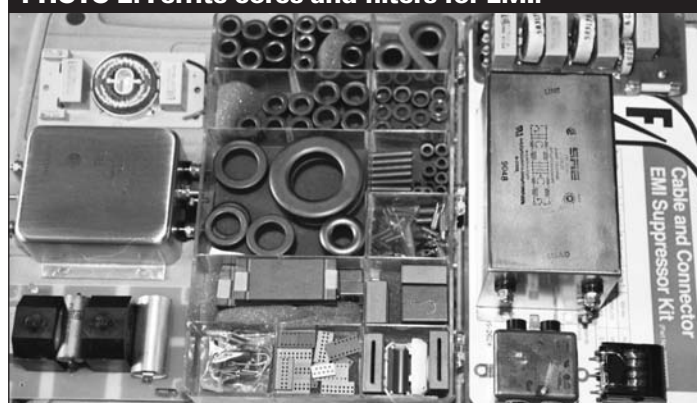
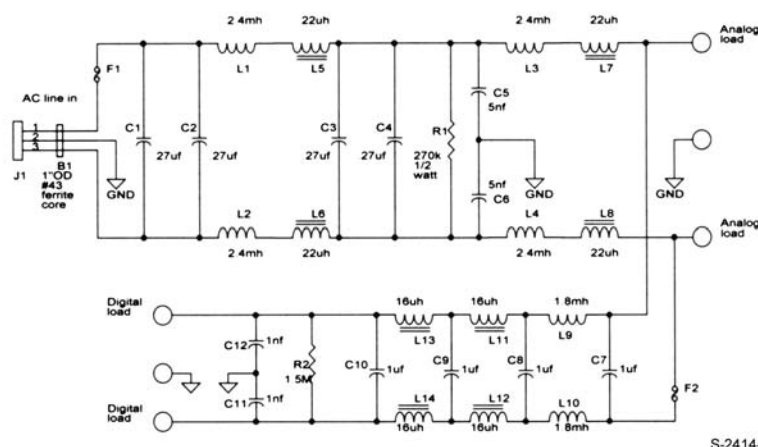


**FIGURE 1: AC EMI filter.**



S-2414-1

your analog from your digital power; otherwise, you just recontaminate your nice clean AC when you plug in your DVD, DAC, or, worse, your PC sound editing system. When installing these filters, make sure the metal case is in good electrical contact with the enclosure. A large ground strap from the filter's GND post to a location where the paint has been removed should ensure this. **aX**

## NOISE SUPPRESSION GUIDELINES

Low frequency fields can be difficult to control, particularly if they are from the AC power transformer. Stray magnetic fields (low frequency) from transformers can cause hum and other annoying noise. Short of a 1" thick lead shield, there are some other effective ways to quell these fields:

- place the transformers in a separate metal enclosure
- use silicon steel shields or shading rings around the transformer
- run cables as close to enclosure sides as possible, and place sensitive circuitry away from power transformers

High-frequency fields can enter a metal enclosure through openings as small as 20mm. This means that all connectors should be metal and holes larger than 20mm be covered in mesh. External cables should be shielded, and internal shielded cables need to be grounded on one end. Enclosure panel thickness should be at least 2mm or more, and heat-sinks should be grounded, if possible, at several points. Use ferrite beads and cores for cables carrying 10kHz or higher (except analog audio signal lines).

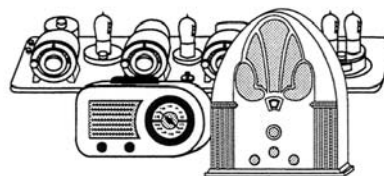
Clean AC cannot be taken for granted, and requires some effort to attain. It has been shown that noise introduced into the power supply can find its way into, and adversely affect, audio signals and ultimately system performance. ■

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# Speaker Damping

This author takes a detailed look at some damping techniques and their effects.

For a rather long time I was convinced that the low output impedance of the modern solid-state amplifiers did a good job of controlling loudspeaker resonance and associated impedance variations. Some rather heated discussions on the advantage of super cables and ultra-high damping factors led to speaker models and speaker/amplifier interface models that cast a shadow on those assumptions. In order to resolve the preceding, the following questions should be answered.

1. What are the audible effects of resonance?
2. How effective is the electrical damping provided by the amplifier?
3. What additional forms of damping are available?
4. What is achievable in the real world?

## RESONANCE AND DAMPING

All cone-type loudspeakers exhibit a resonant point in the frequency response of the unit. The resonance is normally at the low end of the speaker frequency response. Resonance effects of mid-range drivers and tweeters are usually hidden by well-designed crossover networks so that listeners are subjected to only the bass effects. Resonance occurs when a system stores energy at a particular frequency and loses energy at a slow rate after the exciting frequency is removed, allowing the frequency to remain when it should have ceased when the exciting frequency was removed.

You can control the effect to a good extent by inserting a fixed resistance into the system dissipating those lurking frequencies, damping the resonance. The loudspeaker is a mechanical system with a cone/voice

coil combination suspended by the spider and surround. The mass of the cone/voice coil and the spring action of the suspension form a resonant system; friction in the suspension components determines the system losses. If the losses are small compared to the mass of the cone/voice coil, the ratio of stored energy per cycle to energy lost per cycle will be large; this ratio is the quality (Q) of the system.

The audible effects of a relatively high Q speaker resonance are:

1. Notes near the resonant frequency tend to be pulled to the resonant frequency, one note bass.
2. Bass transients tend to be smeared in time and are not well defined, as a result of overhang or ringing combined with a slow attack time due to energy build-up in the resonating components.
3. Bass sounds loose and flabby, overhang prevents speaker from following amplitude variations, and high Q tends to mask small tonal variations.

Table 1 illustrates the number of cycles required for the stored energy to decay by 20dB after the energizing signal is removed. Also shown is the loss after one half cycle. The energizing signal is at the resonant frequency.

Typically bass drivers have Qs of four or more and, if undamped, will exhibit these sonic problems. Reducing the Q to 1.5 will improve things considerably. At a Q of 1.0 the sound will be good and at 0.75 or lower will result in an outstanding bass sound. In order to lower the Q of a bass driver, losses must be introduced into the resonating structure, which is not necessarily a trivial task.

## CONE SPEAKER ELEMENTS

A cone speaker consists of an electrical component—voice coil and magnetic gap—a mechanical component—the cone voice coil assembly—and the suspension components (surround and spider). The air load of the moving cone is also considered a mechanical component. Figures 1 and 2 are equivalent circuits for electrical and mechanical components of a loudspeaker as shown in Olson's *Elements of Acoustical Engineering*. Figure 1 is the voice coil consisting of Rvc (the DC resistance of the coil), Lvc (the voice coil inductance), and Zem (the motional electrical impedance).

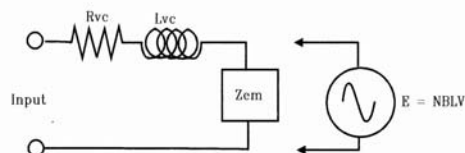
For a 4Ω voice coil, Rvc will be about 3Ω. Zem may be replaced with the voltage generator,  $E = NBLV$ , which represents the back emf generated by the motion of the voice coil for clarity. Figure 2 is the electrical equivalent of the mechanical structure of the speaker. Note that the resistive (friction), inductive (mass), and capacitive (compliance) elements form a series resonant circuit at Fs.

In a series resonance the current is at a maximum and is determined by the DC resistance. Current in an electrical circuit corresponds to velocity in a mechanical structure, in this case the cone and attached voice coil. This velocity is the driving force for the generation of the back emf represented by

TABLE 1: Decay Rate Versus Q.

| Q    | Loss on First Half Cycle | Cycles to -20dB |
|------|--------------------------|-----------------|
| 3    | -1.65dB                  | 6.06            |
| 2.5  | -2.02dB                  | 4.96            |
| 2    | -2.57dB                  | 3.88            |
| 1.5  | -3.52dB                  | 2.84            |
| 1    | -5.3dB                   | 1.9             |
| 0.75 | -6.95dB                  | 1.4             |
| 0.6  | -8.45dB                  | 1.18            |
| 0.5  | -10dB                    | 1.02            |

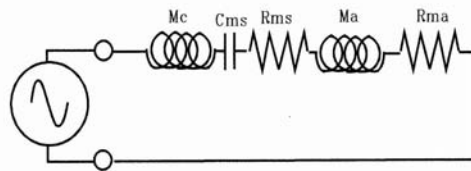
**FIGURE 1: Voice coil equivalent circuit.**



$R_{vc}$  = resistance of the voice coil  
 $L_{vc}$  = inductance of the voice coil  
 $Z_{em}$  = motional electrical impedance =  $(BL)^{-2}/Z_{mt}$   
 $B$  = Airgap flux density in Gauss  
 $L$  = Length of voice coil winding in cm  
 $N$  = Number of turns of the voice coil  
 $V$  = Voice coil velocity  
 $Z_{mt}$  = Total mechanical impedance of the mechanical system,  
 suspension, cone, voice coil and air load

B-2490-1

**FIGURE 2: Electrical equivalent of speaker's mechanical structure.**



$M_c$  = mass of the cone and voice coil  
 $C_{ms}$  = compliance of the suspension, cm/dyne  
 $R_{ms}$  = mechanical resistance of the suspension  
 $M_a$  = mass of air load  
 $R_{ma}$  = mechanical resistance of the air load  
 $Z_{mt} = R_{ms} + R_{ma} + j\omega M_c + j\omega M_a - j/\omega C_{ms}$  where  
 $\omega = 2\pi f$

B-2490-2

$E = NBLV$  of Fig. 1.

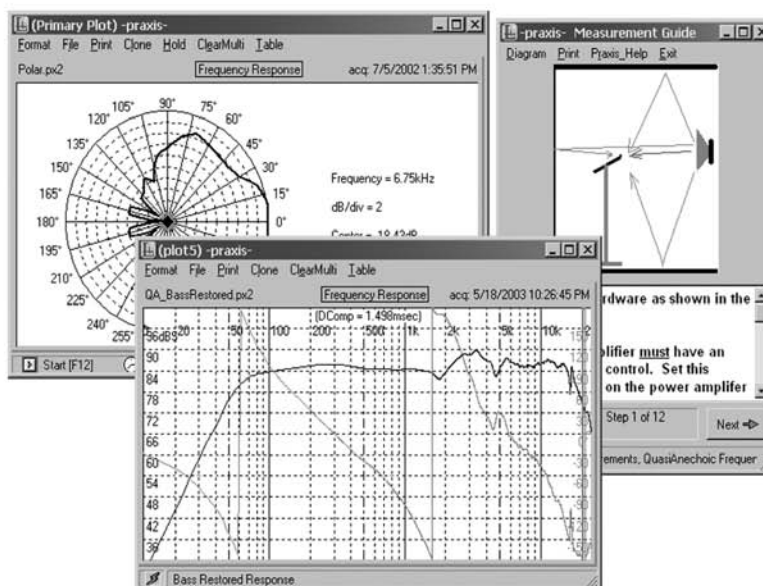
Damping the speaker is accomplished by reducing the resonant velocity of the cone. One possibility is by applying an electrical braking force by shorting the voice coil with an amplifier with a very low output impedance or, as the manufacturers suggest, a

high damping factor. While this is effective to a degree, it is limited by the DC resistance of the voice coil,  $R_{vc}$ , which is in series with the output impedance of the amplifier.  $R_{vc}$  limits the electrical damping effects so that the resonant  $Q$  is reduced to a value of three or more (Table 2).

Examining Fig. 2 reveals an additional damping possibility. The air load provides a mass,  $M_a$ , and a resistance,  $R_{ma}$ , that may be tweaked. Installing the speaker in a sealed cabinet causes the air mass within the cabinet to act as a spring compliance, thus reducing the mass equivalent in-

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ductance and adding series capacitance—compliance of the air within the cabinet—to Cms, which results in the increase of the speaker resonance in the cabinet ( $F_c$ ).

Venting the cabinet through a port tuned to  $F_c$  will change the trapped air to a resistance at the cabinet  $F_c$  and will split  $F_c$  into two peaks, a typical over-coupled response for two circuits tuned to the same frequency. This enhances the low-frequency response as in a tuned port system. Changing the trapped air to a resistance over a range of frequencies centered around  $F_c$  will damp  $F_c$  and remove the resonance effect of the system (see "A Hegeman Subwoofer," Dec. '03 *audioXpress*, p. 12).

## ELECTRICAL DAMPING

Figure 3 details the test configurations used to obtain the data for Table 2. A series resistor is placed in one leg of the cable between the amplifier and the driver. Setting the resistance to 1k $\Omega$  emulates an open circuit across the driver terminals for the constant current test used to measure undamped Q.

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**PHOTO 1: Overstuffed box driven by Hafler DH220.**

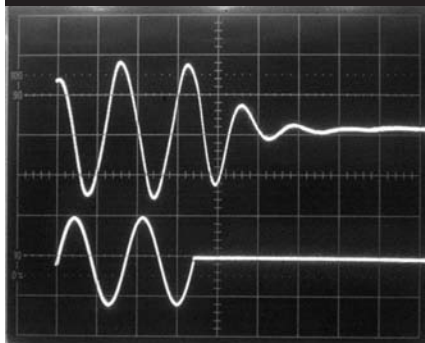
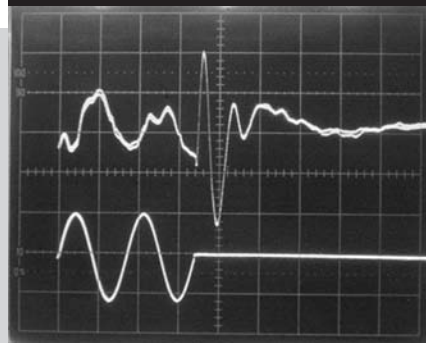


**PHOTO 2: KLH 3-way test results.**



## SERVO CONTROL RESULTS

I have just finished Daniel Ferguson's servo amplifier described in the September 2004 issue of *audioXpress*. Testing was done using the Acoustat subwoofer (Photo 1); it has dual voice coils, with the following results. Photo 4 was taken under the same conditions as Photo 1 except that the servo was incorporated, there was no frequency filtering included so a direct comparison may be made. Excellent control of ringing is shown with an apparent Q of about 0.5 compared to a Q of 1.5 for the unit without the servo. The attack response with the servo shows some improvement at 30Hz, the resonant frequency of the unit, and is almost perfect at 38Hz. The bass is tight, well defined and easy to listen to. Due to the control of the servo it is easy to reach for a volume level that is beyond the system limits causing the servo to lose control. This results in a sonic protest that

immediately catches your attention.

Note that the servo does not change the real Q of the system nor does it prevent energy storage in the system. In reality during the attack period the servo attempts to make the sonic output keep up with the input signal by increasing the applied energy resulting in a greater energy storage and when the stored energy would normally result in output amplitude variations or ring, the servo applies driver energy of an opposing phase to cancel these undesired variations. This is apparent when comparing the decay of the servo output to the decay that is shown in the output waveform of Photo 3. The servo decay is blocky; resembling a half cycle of a low level square wave while that of Photo 3 is an exponential decay. These artifacts are low level and in general do not affect the listening ability of the servo bass which as noted is very good. ■

**PHOTO 3: 10" sub with four tuned pipes.**

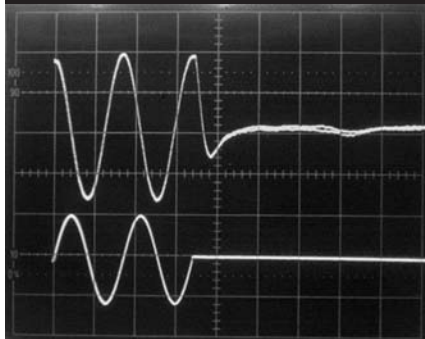
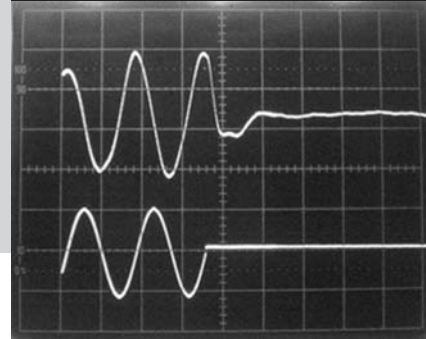


**PHOTO 4: The oscilloscope.**



Placing the AC voltmeter across the driver terminals allows you to read the impedance peak at  $F_s$  as a voltage peak and determine  $F_s$  and the bandwidth. Replacing the 1k $\Omega$  resistor with a 0.24 $\Omega$  resistor emulates a constant voltage drive for the tests used to measure the damped  $Q$ . Connecting a voltmeter across the 0.24 $\Omega$  resistor lets you read the impedance peak at  $F_s$  as a current null and determine the value of  $F_s$  and the bandwidth.

In each case  $Q = F_s/\text{bandwidth}$ . As expected, the damped  $Q$  is significantly lower than the undamped  $Q$ . As the damping efficiency of a shorted voice coil is determined by the current generated by the voice coil, and the current is limited by the voice-coil DC resistance, then the change or delta from undamped  $Q$  to damped  $Q$  should be inversely proportional to the voice-coil DC resistance,  $R_{vc}$ . The apparent challenge to this argument is the 6" midrange, but starting with a  $Q$  of 19.7 very little electrical damping is required to reduce the  $Q$  to 17.4. The remainder of the group follows nicely, with the JBL leading with a delta of 3.5

and the RFD 1208 trailing with a delta of 1.2.

The RFD 1208 ends with the lowest damped  $Q$  of 3.7, but it started with the lowest undamped  $Q$  of 4.9 due to high frictional losses in the surround and spider. Clearly the electrical damping provided by an amplifier results in a significant reduction in driver  $Q$ , but only down to a value of three or so and not down to the magic area below one. Several methods to decrease the value of driver  $Q$  and extend bass response follow:

## HIDE IT

The car audio people go for small sealed enclosures with volumes of  $\frac{1}{2}$  to  $\frac{1}{4}$   $V_{as}$ . The small enclosure volume causes the driver resonance to increase by a factor of 1.7 to 2.2 above  $F_s$  and, you hope, above the operating

**FIGURE 3: Test setup to measure resonant frequency,  $F_r$ , and bandwidth.**

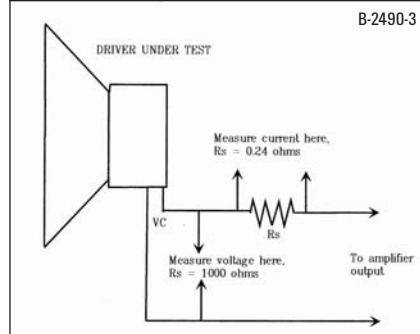


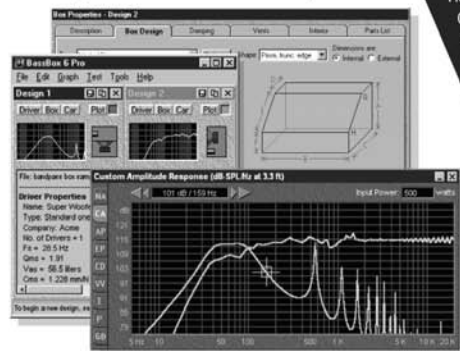
**TABLE 2: Electrical Control of Driver  $Q$ .**

| Driver                           | $R_{vc}$     | $Q$ , constant I | $Q$ , constant E | Delta |
|----------------------------------|--------------|------------------|------------------|-------|
| MCM 55-2330 12"                  | 6.5          | 6.6              | 4.5              | 2.1   |
| Realistic 40-1026* 12"           | 6.4 $\Omega$ | 7.7              | 4.8              | 2.9   |
| Optimus 40-1028* 10"             | 6.7 $\Omega$ | 7                | 5.4              | 1.6   |
| Rockford Fosgate RFD 1208 8"     | 7.2 $\Omega$ | 4.9              | 3.7              | 1.2   |
| Dynamic Speaker Corp 6" midrange | 7.3 $\Omega$ | 19.7             | 17.4             | 2.3   |
| JBL 125A 10"                     | 5.2 $\Omega$ | 7.8              | 4.3              | 3.5   |

\*The Radio Shack drivers with these designations are subject, like all RS drivers, to wide characteristic variations.

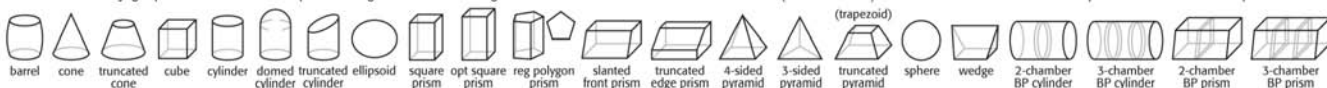
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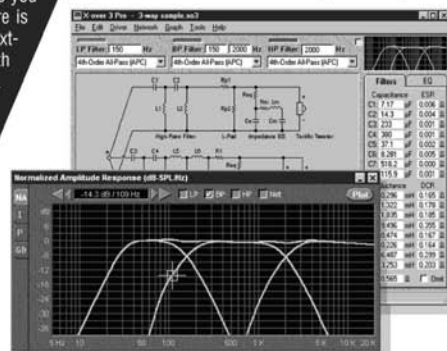
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range of the subwoofer. The extreme stiffness of the small trapped air volume also tends to swamp some of the resonance peak and makes the unit very hard to drive. They are power hungry.

The ultra long-throw woofers that make reasonably sized subwoofers possible were developed for car audio, so thanks, guys.

## SERVO CONTROL

Controlling the cone motion by a separate feedback loop servo has been a Holy Grail of speaker builders since the early thirties. There exist several that provide excellent control at a stellar price. The servo unit is rather complicated and when driven to the driver limits may lose control, which results in highly unusual sounds. An excellent pair of articles, "A Servo Dual Voice Coil Subwoofer" and "A Dual Voice Coil Servo Subwoofer System" (Nov. '03 and Sept. '04 *audioXpress*), both by Daniel L. Ferguson, give a good discussion of the ins and outs\* of servo control and present design and construction detail to enable the home builder to try this

for him or herself.

## RESISTIVE LOADING

Using the air load as a resistive element has been tried in several forms. Transmission-line speakers use a quarter-wave line tuned to the resonant frequency of the driver when installed in a box of the same volume as the transmission line. The cross-sectional area of the line is equal to the piston area of the driver. As the air in the line becomes more resistive and less of a spring at resonance, the actual driver resonance is lowered.

The line is typically stuffed with fiber to reduce higher-order line resonance and increase resistance. This results in slowing the sound through the line by 10% to 20%, lowering the tuned frequency of the line. Fine-tuning stuffing and possibly line length will result in an alignment of the driver resonance and the line resonance.

The finished product should have a very smooth impedance curve with very small resonant peaks and a rather gentle rolloff in bass response below the line resonance resulting in

a well-controlled low-end response. Transmission-line speakers generally use wide-range 6" to 8" drivers with limited pumping volume, which limits their low end to 50Hz or higher. To achieve a sound pressure of 90dbi at one meter at 50Hz requires the displacement of 214cm<sup>3</sup> of air per stroke and at 25Hz, 856cm<sup>3</sup> is required.

Overstuffing a closed box will increase the resistance of the air load and reduce the driver Q to some extent. *Photo 1* illustrates an overstuffed box driven by a Hafler DH220 operating in the bridged mode. The Q is approximately 1.5 as illustrated by the ringing of the driver after the drive signal was turned off.

The bottom trace is the 30Hz drive and the top trace is from a microphone pickup located 2" above the driver cone. This speaker is an older 10" Acoustat bass unit used with their electrostatic sticks. The sound is not bad but slightly muddy.

The tuned port systems result in a resistive air load at the tuned frequency of the port, matched to the driver, but overcoupling results in two resonant peaks. One peak is below the tuned frequency of the port and one above it. The lower peak will result in a bass extension; however, controlling the resonance is difficult, as illustrated by *Photo 2*.

The conditions are the same as for *Photo 1*—note that both resonant peaks have been energized and appear in the output. Also close examination shows an intermodulation sum. The speaker is loud but very muddy and undefined in bass. The speaker tested was a popular KLH 3-way using a 12" bass driver with a crossover at 900Hz, well above the 30Hz test frequency.

In 1968 A. Stewart Hegeman introduced his model 1, which included a point source launch and quarter-wave tuned pipes to control the resonance of the bass/midrange driver. He used six quarter-wave length tuned pipes opening into a small plenum below the driver with the opposing ends closed. At resonance each pipe presented the driver a resistive air load whose magnitude was dependent upon the Q of the pipe. By tuning the pipes around the resonant frequency of the driver,

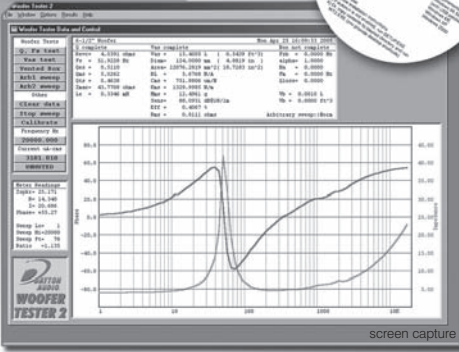


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they formed a bandpass resistive load to the driver.

Because the open ends of the pipes were at the point of highest particle velocity, they were easily adjusted in Q to provide a smooth resistive load to the driver that easily went from one-half octave or so below to one octave above the driver resonance, resulting in a well-defined and articulate bass presentation. Since the pipes are of a higher Q and are situated so that the open end loads the driver, they require a total pipe cross-section area of only 50% to 80% of the cone piston area, compared to 100% of the transmission line, which results in a smaller box size.

*Photo 3* shows the output of a 10" subwoofer using four tuned pipes. Again the conditions are those of *Photo 1*. Note that the small half cycle of ring is about 10dB down from the peak, indicating a Q of about 0.5. Also note that the peak of the first half cycle is approximately 90% of the second peak, as opposed to *Photo 1*, where the first peak is only 80% of the second.

The Q of this unit when driven by a constant-current source is 0.8; the low output impedance of the amplifier reduces the Q to the 0.5 level. This is the subwoofer described in Dec. '03 *audioXpress*.

Have the initial questions been answered? I believe so.

1. Bass clarity depends upon a low Q bass driver system.
2. The effect of the electrical damping provided by the amplifier is dependent upon the DC resistance of the voice coil and cannot fully damp resonance effects; a final Q of about 3 seems to be the limit.
3. Several methods of providing additional damping are available.
4. Near perfect damping characteristics may be achieved, but they are expensive and/or labor-intensive.

## PHOTOS

The output signals of *Photos 1* and *3* exhibit a 90° phase lead compared to the input signal. This is not a microphone artifact, using the same microphone it has been measured on drivers with

## TEST EQUIPMENT

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0.24Ω resistor, 2 ea. 0.47Ω resistors in parallel  
1kΩ resistor, carbon film

resonance frequencies of 30Hz, 42Hz, 78Hz, and 134Hz. In all cases it disappears when the signal frequency is increased by one to one-and-one-half octaves above resonance. It seems to be a characteristic of cone drivers, even in a well-damped system.

Now where do I put that subwoofer?  
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*[\*Puns, intended or not, are welcome here.—Ed.]*



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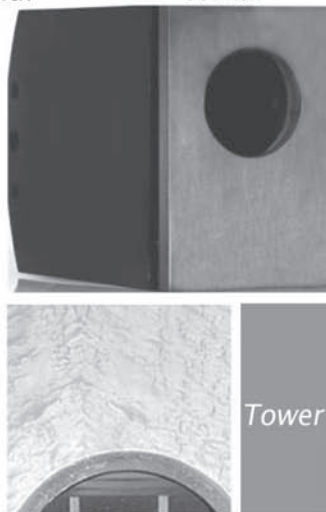
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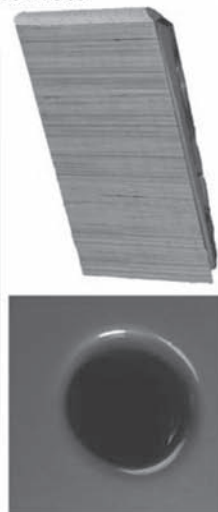
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# A Full Bridge Output Amp

By Monny Nisel

This author shows how you can use a four-tube bridge arrangement in amp design.

PHOTO 1: The completed amp.



This project started as a result of the circlotron amplifier I built several years ago. Although I liked the pleasant sound of the circlotron, I was annoyed by its major and well-known drawback, namely, the use of two independent power supplies for each channel. Thinking about a simplified solution, I tried to figure out what was different in the circlotron arrangement that makes it sound so good.

Apart from benefiting from the cathode-follower output loading and the low-impedance power stage, there is one more element that seems to go unnoticed but is quite significant, in my opinion: the total absence of DC through the primary winding of the output transformer (i.e., the DC supply of the output tubes is not flowing through the transformer winding). This is a completely different concept from most of the classic arrangements. I intended to reproduce those conditions in a simpler manner, and I first thought

about the single-ended push-pull schematics with the two tubes in series supplied from a single source.

Unfortunately, this arrangement needs a double power supply (although not two independent ones), or a series capacitor, or a capacitor divider to hook up the load. This idea did not look too appealing. At this point I figured I could try to use two single-ended push-pull circuits connected in parallel on the same DC supply with the load connected to both median points of the two branches. Well, it did not take me long to figure out I was not the first to have this idea.

I remembered seeing something similar in an old *Glass Audio* magazine. The article is called "Linear Output Stages," by Graeme J. Cohen (GA 6/95, p. 34).

A search on the Internet revealed a description of an amplifier using two 6AS7 tubes in the same arrangement and capable of 9W class-A out-

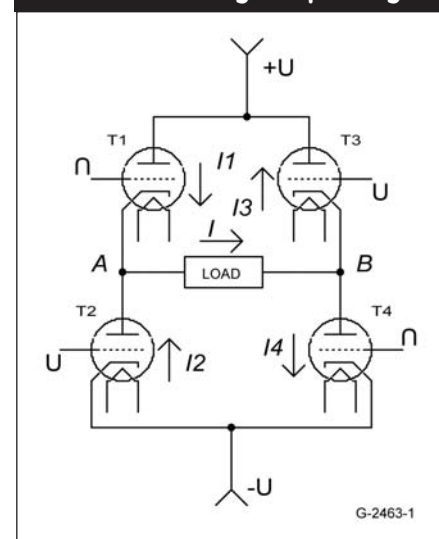
put power to be used with a 250W impedance speaker. This description by Kerim Onder was published in the October 1953 issue of *JAES*. As 250W speakers are history and 9W from two 6AS7 tubes seemed quite low, I figured that I should try a different approach in order to upgrade the initial concept to higher output power as well as to the use of regular 8W speakers.

## FULL-BRIDGE OUTPUT-STAGE OPERATION

I called this arrangement full bridge to differentiate it from the circlotron, which is sometimes referred to as the circular bridge or simply bridge amplifier. It seems there was not much interest in any commercial audio amplifier using this arrangement, and for some good reasons: low plate efficiency, use of minimum four tubes in one channel output stage, and high supply voltage for the tubes in series.

However, most of these difficulties can be overcome by using a double triode such as the 6AS7. The configuration and operation of the full-bridge output stage is shown in *Fig. 1*. Two tubes are in series, on each branch, as

FIGURE 1: Full-bridge output stage.



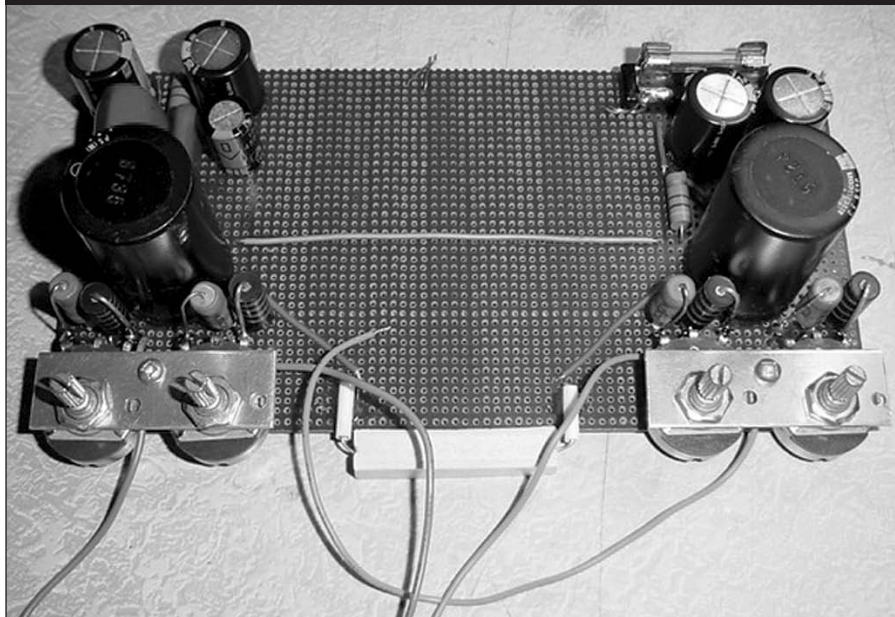
in a single-ended push-pull, with the two branches in parallel on the power supply. The load is connected between the midpoints of the two branches.

All conditions being equal (i.e., tubes matched and appropriately biased and with no signal at the grids), the DC voltages will be the same (approximately  $U/2$ ) at the two middle points A and B. As a consequence, no DC current is flowing through the load.

Applying the signal to the stage is somewhat different from any previously known arrangement. Here is a simple explanation of the way the stage works. The bridge operates with so-called "diagonal phasing," meaning that you must apply one phase of the signal to tubes T1 and T4 and the opposite phase to tubes T2 and T3 as shown in Fig. 1. As the positive period of the signal is applied to the grid of T1, the current, as well as the voltage of the tube cathode (point A), increases (I1). The negative period applied to the grid of T2 decreases the tube's current (I2), keeping the potential of the tube's plate high (point A).

Practically, the effects of the opposite

PHOTO 2: Power supply assembly.



phase application are cumulating in point A, increasing the potential of this point. The same type of operation, but with reverse signal phasing applied to the right branch (T3 and T4), leads to the decrease of the voltage in point

B. As a consequence of the voltage unbalance of the two points (A and B), a current—I—following the signal flows through the load.

From the signal point of view, the two tubes on each branch supply the

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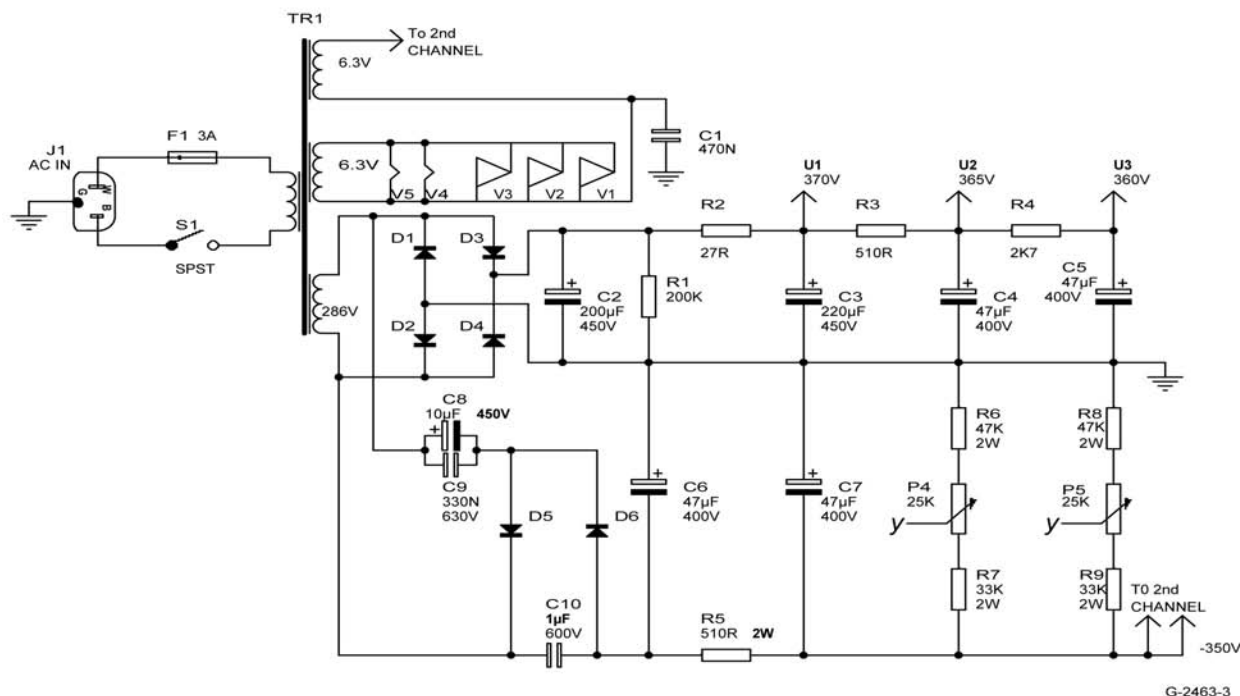
triode (V1) and provides the signal level to drive the first phase splitter, using a 12AU7 tube (V2) wired in the long-tail configuration. The output of the first

phase splitter drives the two second splitters using the two triodes of V3 in an unusual cathodyne configuration.

These two splitters also act as driv-

ers, each triode providing the signal with the right phasing and level to a pair of output tubes—V4-1, V4-2, and V5-1, V5-2, respectively. The bridge of the

**FIGURE 3: Power supply—first version.**



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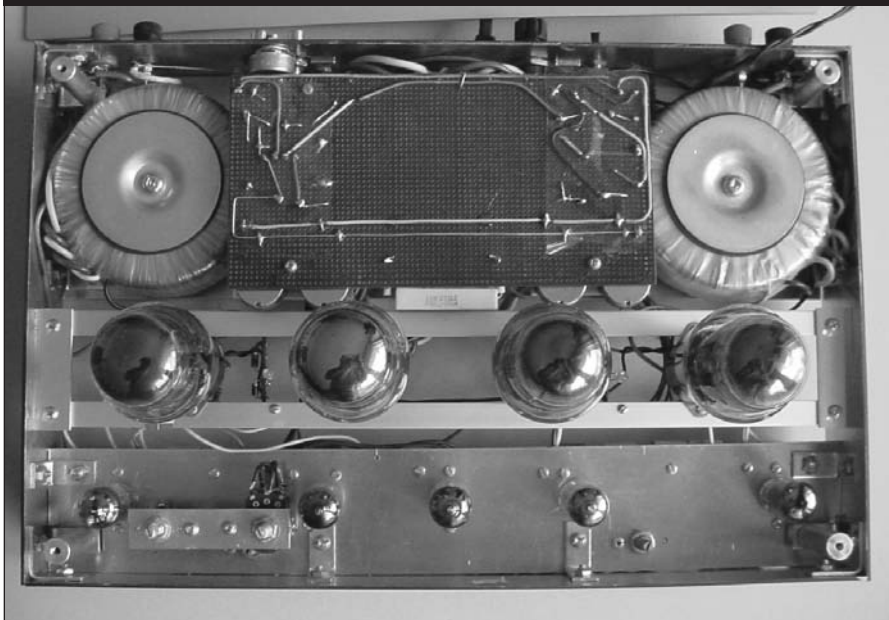
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PHOTO 4: Overhead view.



output tubes is redrawn to fit easier with the balance of the schematics. The use of the 6AS7 tube in the output stage came as a natural choice because it contains two completely separate tubes in the same envelope, has a low plate resistance and good sonic quality, and is easily available at a low cost.

I used the low-mu 12AU7 double triode in the two splitter stages. Apart

from getting higher drive current for the power stage, the 12AU7 is an inexpensive tube, which you can find even in a NOS version for a few dollars—this counts when you need four of them for the stereo version of the amplifier. I added an optional negative feedback circuit with two steps of 3 and 6dB between the output and input of the amplifier.

Though my personal preference was to listen to the amplifier without any feedback, some listeners preferred the sound with a certain degree of it. The three-position switch facilitates those preferences. I opted for a small amount of feedback, but I noticed quite a significant extension of the bandwidth and drop in distortion. Potentiometer P2 sets the balance between the two channels.

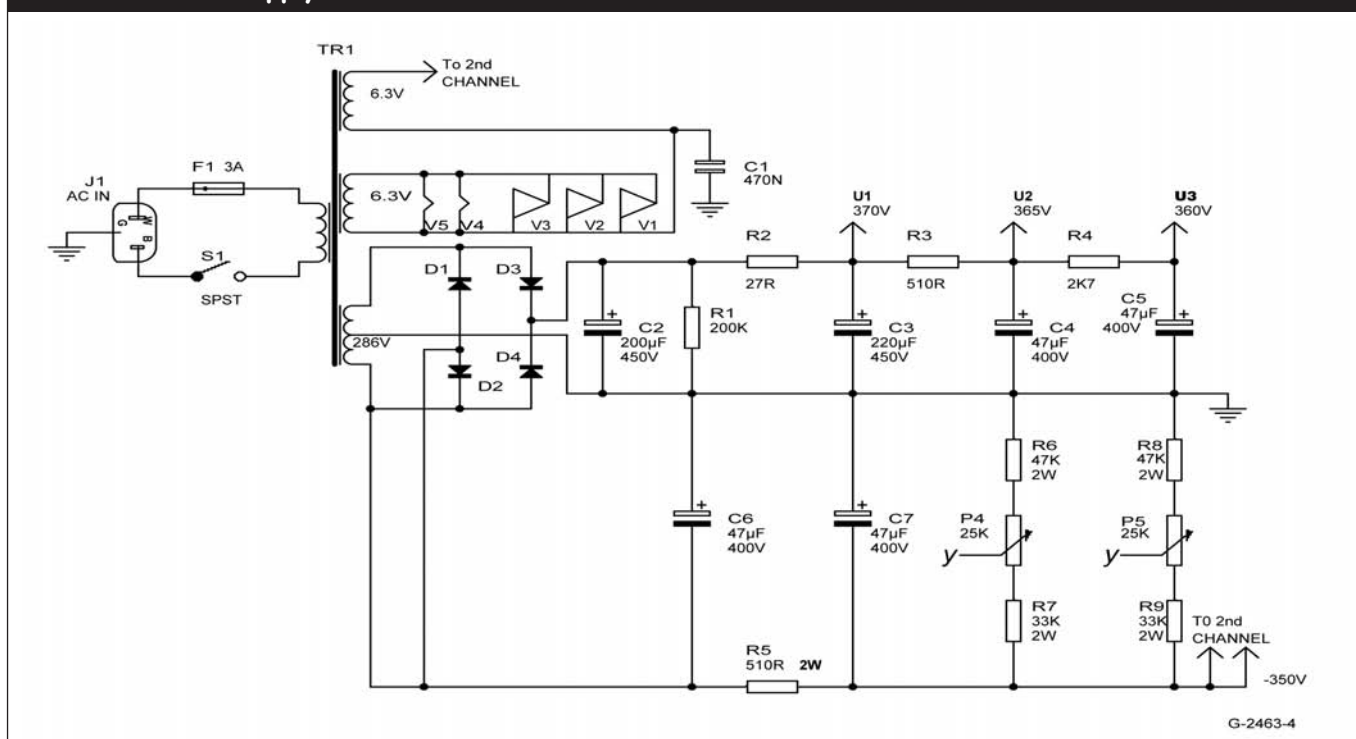
## THE GAIN STAGES

The amplifier's total gain without feedback is 45dB, almost all of it achieved in the first two stages, the input stage, and the long-tail phase splitter. The coupling between these two stages is direct without any capacitors. There are practically only two coupling capacitors in the entire schematics of the amplifier. The voltage in the junction point should be around 80V DC. To drive the amplifier to full output power, you need 0.5V RMS; the first stage (V1) develops 14V RMS and the long-tail phase splitter 75V RMS on each output.

## THE SECOND SPLITTER/DRIVER

Contrary to the regular cathodyne phase splitter, the one used to drive each of the output tubes has quite a different configuration. The load in the

FIGURE 4: Power supply—second version.



cathode is the 100k $\Omega$  resistor (R12 and R13), while in the plate it is the 270k $\Omega$  resistor (R14 and R15) connected directly to the positive rail. I initially started with the same load 100k $\Omega$  in the plate circuit connected to the middle point of the output power tubes (A and B). This arrangement was supposed to provide better stability through negative feedback from the output stage, as I found in some recommendations.

However, I could not reach more than 14 to 15W this way because the upper output tube operating as a cathode follower did not get enough drive, and after I tried several solutions I came up with the actual one. This way the drive voltage is higher with approximately 10%, and I could reach 18W of output power. I did not notice any instability or measurable increase of distortion.

The resistors (R14, R15) set up the initial bias of the upper output tube, bias which subsequently follows closely the adjustable bias of the lower output tube. The value of the bias resistor of the cathodyne (R9, R11) is critical in setting the right DC balance between

the two series output tubes. In case somebody wishes to try my initial arrangement by replacing R14, R15 with 100k $\Omega$  resistors connected to the median points A and B, the value of R9, R11 should be 12k $\Omega$ .

## OUTPUT STAGE

When I began experimenting with the output stage, I started by using automatic bias as recommended for the 6AS7-type tube supplying the tubes through heavy cathode resistors. The results were disappointing: an important drop in the DC supply voltage, a lot of heat dissipated from the resistors, and, most important, the power output was less than 10W, all of which I did not like. Then I decided to try the automatic bias. While I was unaware of any example in the literature of the 6AS7 tube biased this way, I later found out that some designs used it despite the RCA warning against it.

I first tried this arrangement in my circlotron 6AS7 and, though it worked, the DC supply voltage of the tubes was much lower, and I tested it for only a short time. Being quite concerned with

what might happen with the tubes in series supplied at close to 400V DC and fixed bias, I prepared myself with a bunch of used but good 6080s meant to be sacrificed. Fortunately, things went smoothly: apart from a couple of tubes which I damaged by mistake, most escaped and seemed to work well with fixed bias. The two tubes in series on each branch are supplied with 370V, or approximately 185V per tube when all is balanced.

The average bias voltage of -85 to 90V applied to the grid of the lower tube allows for a quiescent current of 55mA and a power dissipation of just over 10W/tube. This power is quite close to the maximum dissipation power of 13W/tube. I like to run it this high because the sound quality is much better and the price to pay—the shortening of the tube's life—is not too high due to the low cost of the 6AS7. However, it is your choice to run the tube at lower current by adjusting the bias voltage accordingly. With this bias adjustment and an average drive signal of 55V RMS, the output stage operates in class AB and delivers an output



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power of 18W before clipping.

## THE POWER SUPPLY

As Figs. 3 and 4 show, the high voltage DC power supply is straightforward. I used a toroidal power transformer, which I ordered at SumR for a very convenient price. Because I initially planned to use automatic bias, I did not provide for additional voltage to use for fixed bias; so when I switched to it, I needed something to avoid adding a new transformer. I came up with a simple voltage doubler, connected to the same high voltage AC winding on which I tapped the negative output to

get roughly -350V with respect to the negative rail of the DC high voltage supplying the tubes.

I later ordered another power transformer with a double high voltage winding from SumR, which I used on another sample of the amplifier. As you can see from the alternate schematics, the bias voltage is simply taken from the high voltage winding in a standard configuration. Both versions work well. The bias adjusting circuit provides between 70 and 100V, adjustable through potentiometers P4 and P5. The power transformer—in both versions—is capable of supplying the stereo configuration of the amplifier.

### PARTS LIST

#### 1—AMPLIFIER

|             |                     |      |                       |
|-------------|---------------------|------|-----------------------|
| R1          | 1200R               | 1W   | cc (carbon composite) |
| R2          | 270k                | 1W   | cc                    |
| R3          | 1k                  | 1W   | cc                    |
| R4, R8, R10 | 1M                  | 1W   | cc                    |
| R5          | 22k                 | 2W   | cc                    |
| R6, R7      | 47k                 | 2W   | cc                    |
| R19, R11    | 10k                 | 1W   | cc                    |
| R12, R13    | 100k                | 1W   | cc                    |
| R14, R15    | 270k                | 1W   | cc                    |
| R16, R17    | 5.1R                | 0.5W | cc                    |
| R18         | 12k                 | 1W   | cc                    |
| R19         | 4.2k                | 1W   | cc                    |
| R20         | 200R                | 1W   | cc                    |
| C1, C2, C3  | 0.47μF              | 400V | polypropylene film    |
| P1          | 100k                |      | potentiometer log.    |
| P2          | 250k                |      | potentiometer log.    |
| P3          | 25k                 |      | potentiometer lin.    |
| V1          | 12AT7               | tube |                       |
| V2, V3      | 12AU7               | tube |                       |
| V4, V5      | 6AS7G (6AS7GA, 6080 | tube |                       |

#### TR1 – output transformer

Md. # Rx 01920302  
450/8Ω from SumR

#### 2 – POWER SUPPLY both versions

##### TR1 Power transformer: Md # RC0300 055 B, Make SumR

Primary: 2 × 115V  
Secondary: 2 × 6.3V, 7A  
286V, 0.55A

or Md. # RC0300 055 5, Make SumR  
Primary: 2 × 115V  
Secondary: 2 × 6.3V, 7A  
2 × 286V, 280mA

|        |         |      |                    |
|--------|---------|------|--------------------|
| D1-D6  | 1N 5408 | 103A | diode rectifier    |
| R1     | 200k    | 2W   | cc                 |
| R2     | 27R     | 10W  | wire wound         |
| R3, R5 | 510R    | 2W   | cc                 |
| R4     | 27k     | 2W   | cc                 |
| R6, R8 | 47k     | 2W   | cc                 |
| R7, R9 | 33k     | 2W   | cc                 |
| P4, P5 | 25k     | 1W   | potentiometer lin. |
| C1     | 0.47μF  | 400V | polypropylene foil |
| C2, C3 | 200μF   | 450V | electrolytic       |
| C4-C7  | 47μF    | 400V |                    |
| C8     | 10μF    | 450V |                    |
| C9     | 0.33μF  | 630V | polypropylene foil |
| C10    | 1μF     | 630V | polypropylene foil |

straightforward type; the high voltage winding is either single for use with a bridge rectifier, or double for two separate rectifier diodes. Two 6.3V 7.5A heater windings each supply one-half of the tubes in the stereo version.

A second reason for using toroidal transformers was the output transformer's simple construction. The transformer has an impedance ratio of 450/8Ω. With one single interleaving—the secondary is wound between two halves of the primary—I received an almost flat response between 25Hz and 100kHz when I drove the primary from a generator with the secondary connected to an 8W load.

The low primary impedance helped a lot, but an important factor is the tight coupling specific to toroidal transformers. My main goal when choosing the transformers was to minimize the impact they have on the cost of the amplifier without significant compromises on the quality, thus making the amplifier more affordable to more people. With this output transformer the ±3dB frequency range of the amplifier extends between 20Hz and 50kHz at full 18W output power and without negative feedback. The sound is transparent and airy especially in the middle and higher range, voices and instruments showing an unusual presence. However, for those used to the sound of heavy damped pentode or beam tube output stages, the sound may be disappointing for the lack of heavy lower bass.

## AMPLIFIER ASSEMBLY

I assembled the amplifier in a stereo version, on a somewhat unusual “mecano toys”-type chassis made from aluminum sheet strips, held together with metal angles and screws. I purposely did it this way so I could fit it with different face plates and change the look of the amplifier. The photos show a simple aluminum face plate, which I do not recommend because it was so time-consuming. Instead, you could use a Hammond type 17 × 10 × 3” chassis, on which you install the components in a classical way (i.e., transformers and tubes on top and all the smaller components under the chassis).

I installed the power supply and negative bias circuitry components on a vector board so I could fit them in the

## THE TRANSFORMERS

The transformers I used in this amplifier are all toroidal types, which I chose for several reasons. The most important was the opportunity to find a manufacturer called Sumner Technologies in Etobicoke, Ontario, who agreed to build me the transformers, without any preconditions and for a price that beat any of the well-known manufacturers. As it happened, Sumner Technologies is a manufacturer of toroidal transformers. The president of the company, Richard Sumner—a businessman with a busy schedule—found the time to listen to my requests and changes, and if the transformers came out right I must credit Richard.

The power transformer is a

small space around the power transformer. When using a standard type chassis, solder the components, both for the power supply/bias voltage and the rest of the circuitry, on terminal strips installed close to the main components they are related to (power transformer, tube sockets).

The stereo version of the amplifier includes eight potentiometers, which you must consider when planning a mechanical layout. Install balancing potentiometer P2 close to the main input potentiometer or close to the input RCA jacks. Mount potentiometers P3 (and P1 for the other channel not shown) in the proximity of the long-tail splitter. Place the bias potentiometers P5, P6, as well as adjacent resistors, next to the output tubes they supply.

The output tubes dissipate over 35W of power each—a lot of heat for the small envelope of 6AS7 GA or 6080. For this reason spread out the tubes and eventually drill ventilation holes around the sockets.

With the components installed, I wired the amplifier starting with the transformers, then the heaters of the

output tubes (which require at least #18 wire gauge) and advanced to the first stages. I wired small components using terminal strips.

## AMPLIFIER ADJUSTMENTS

After wiring and with the right fuses in place—but without the tubes—I measured to make sure the voltages were right. In order to prevent the DC voltage from rising to dangerous peak levels on the power supply capacitors without the tubes, I connected a temporary load made up of a 5kΩ 20W resistor to the output of the power supply. When using this option of loading the high voltage, be aware that the resistor becomes extremely hot, so avoid touching it and isolate it from surrounding objects. A better way of powering the amplifier is through a Variac autotransformer and watching until DC voltage reaches 370 to 380V.

I strongly recommend measuring the voltages at the output power tube sockets and paying special attention to the presence of the negative bias voltage at the tube's grids. Before plugging in the tubes, turn the bias potentiometers so

you have the maximum negative voltage at the grids. To set the quiescent current of 55mA, you need to adjust the bias potentiometers P5, P6 until you measure 275mV on each of the cathode resistors R16, R17. You must perform the operation a few times, going back and forth from one tube to the other until you reach the right voltage drop on each of the two resistors.

The two 5W/0.5W, R16 and R17, resistors also act as fuses in case of a sudden and strong overloading or shorting of the tubes. After setting the quiescent current, it is important to check the balance of the output stage. You can start by measuring the bias voltage to check how the tubes match; a difference of 2-3V indicates a good match.

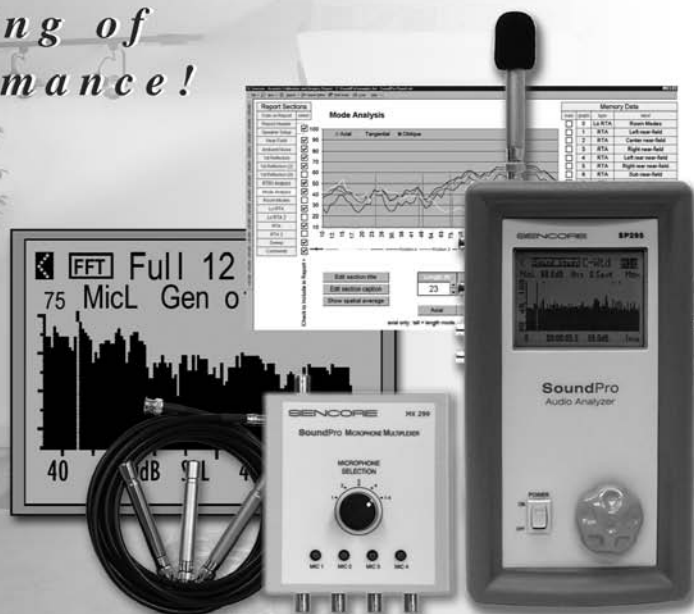
Next, disconnect one of the output transformer legs from the median point of the two tubes and measure the DC voltage across each of the output tubes. You should find half of the total DC supply voltage for a good balance. Again, a difference of 3-4V is acceptable.

If the difference is much higher, the culprit could be the lack of match-

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ing between the two halves of the output tubes or of the cathodyne splitter halves (V3). Even resistors R9, R11 and R14, R15 should be matched as closely as possible. The balancing of the output stage is more demanding in this type of amplifier if you plan to squeeze a few more watts of undistorted power. The last adjustment, which is straightforward, is the AC balance of the long-tail splitter through potentiometer P3.

## THE DISTORTION PROBLEM

According to the articles previously mentioned, this type of output stage is one of the most linear with distortion below 1%. However, there is a difference between those amplifiers as per class of operation and power output. This amplifier is operating in class AB and outputs double the power of Kerim Onder's amp, which works in class A only.

Moreover, this design uses an output transformer, an additional source of distortion. With two well-matched tubes, the stage closely balanced, and no negative feedback applied, the distortion measured at 1kHz was 1.4% at full 18W output power; the distortion drops to 0.2% at 1W output power.

However, there are a few ways to decrease the distortion. The first (which I already mentioned) is by applying the negative feedback. With 6dB of negative feedback, the distortion drops almost a full percentage. Another method, suggested by Graeme J. Cohen, was to insert a small resistor into the positive DC rail supplying the two upper output tubes. According to the author, the resistor helps to balance the mutual conductance of the two tubes, thus contributing to reduce the second-order harmonics. The article does not indicate any value for the resistor so I tried a few myself.

A 100Ω resistor will reduce distortion by more than half a percentage point but will also drop close to 2W of output power. Another method I tried was to adjust P3 for the lowest distortion; the results were almost similar to the resistor method.

Because I worked hard for every watt of available power from this amplifier, I was reluctant to lose any of them, so I stayed with the negative feedback only. In one switch position

with the feedback network disconnected, there was no distortion reduction procedure applied. This was, according to my own taste, the position with the best sound.

## A FEW MORE PRACTICAL NOTES

The following items, based on many hours of experimenting, will make it easier for you to build the amplifier.

- **The DC high voltage supply of the amplifier.** As previously mentioned, by using the SumR mains transformer, the DC high voltage in the stereo version will reach a maximum of 380V (or 190V per tube). I found that it is not safe to go over this voltage because of the thermal runaway tendency of the output tubes. This seems to be more evident with the 6080 version of the tube.

I mention this problem because of the existence on the market (or in hobbyist junk boxes) of over 300V AC winding transformers, which some may be tempted to use. If so, you need to lower the DC voltage to a safe level. My main voltage is a couple of volts lower than normal—the DC high voltage is 370V and the amp works well.

- **The DC voltage on V2-1 grid (point X).** Although most of the time the long-tail splitter will keep its cathode approximately 9V higher than the grid, I found that the voltage at the grid should be around 80V DC ( $\pm 3-4V$ ) for the optimal operation of the stage. This voltage, being a function of the plate current of V1-1, may vary with different samples of the tube. To correct the problem you may need to change the tube or slightly alter the bias resistor. I could have solved this problem by using capacitive coupling between the stages, but I preferred to have one less capacitor in the signal path.

- **Matching the output tubes and balancing the output stage.** The 6AS7/6080—in many respects a beautiful tube—is particularly known for having quite a wide spread of parameters from one sample to another, even in tubes from the same batch. The problem of matching is further compounded in this amplifier by the need to match four tubes, two being located in the same envelope. The best

thing is to order two pairs of matched tubes from a supplier, but I imagine many hobbyists would like to use their own stock as I did. Because the test conditions were far apart from the operating conditions, and thus quite impossible to match the tubes in my tube testers, I resorted to matching directly in the amplifier.

When adjusting for the same quiescent current in the two tubes, I sometimes found differences of 10-15% in the bias voltage. In such cases I prefer to adjust for 3-5% difference in current, thus lowering the unbalance of the bias voltage. Also, you get an approximate idea of the mutual conductance by measuring the AC voltage on the cathode resistors of the output tubes when testing the amplifier for output power. Differences of 10% are not unusual. Even when you cannot match the tubes well, you can still expect 14-15W of output power before clipping.

- **Identical conditions for test.** The 6AS7/6080 tube takes quite a long time until it reaches thermal stability after you switch on the amplifier. I found it is best to wait 5-10 minutes before attempting any tests or adjustments.

Another factor to consider is the mains voltage. A drop of a couple of volts in the AC mains can translate to close to a 10V drop in the DC high voltage supplying the tubes, and this

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can mean quite a change in previously measured values. You must be aware of that factor so you don't become frustrated when you find that the output power is 1W less or the bias voltage is 5V higher than you measured a few days ago. I mention this because it happened to me all the time, and I would become frustrated even though I knew where it is coming from. My mains supply varies between 112 and 118V.

- **The amplifier test under full power.** When doing tests under full output power, the tubes overheat and the test should not last for more than 20 to 30 seconds at a time with appropriate cooling breaks.

- **The heater-induced hum problem.** Due to the configuration of the mains transformer, with its single heater winding per channel, as well as the amplifier arrangement, I preferred to leave the heater floating with respect to ground. To minimize hum I connected the heater winding to ground through a 0.47 $\mu$ F capacitor, which seemed to work well. I measured a barely audible 2mV at output with volume control at minimum hum. When experiencing hum from an electrically noisy mains, connect a 0.1 $\mu$ F/600V polypropylene dielectric capacitor between ground and each of the two 117V transformer primary wires.

- **Safety.** Voltages in the amplifier are lethal, so be very cautious when you work on it. Also beware of briefly turning the amplifier on or off; the heaters may not have the time to warm up while the capacitors have time to fully charge up. Modern electrolytic capacitors can keep the charge for many days.

## A CONFESSION

I built my first version of the amplifier with low voltage power transformers in the output stage. Though I knew the output transformer should have a low primary impedance, I could not find anything available on the market, and I did not want to place a firm order before determining the optimal value of this impedance.

As I started experimenting with this project, I tried six different output impedances from 320 to 640 $\Omega$  and record-

ed the highest output power at 450 $\Omega$ . I decided to try a pair of 120/16V -150W toroidal transformers that I found in my junk box.

This transformer reflects the 8 $\Omega$  speaker impedance as a 450 $\Omega$  load seen by the amplifier. Though the measured results were somewhat at the limit of decency, the sound was good enough to encourage me to complete the amplifier in a stereo version using these transformers, as well as to order—not too long afterwards—the right transformers from SumR. I am still listening to this pleasant-sounding amplifier from time to time.

So why this confession? I know many amplifier hobbyists have well-garnished junk boxes and not much money, so there is a choice, though not the best one. *aX*

*Monny Nisel is designing and building tube amplifiers as a hobby after retiring from the aerospace industry where he worked as an electrical and instrumentation engineer. He likes to try mostly projects which fall outside the beaten path.*

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# Five Speaker Myths Undone

By Mark Sanfilippo

The worldwide Internet is a critical tool in this information age. But it can also lead to widespread misconceptions, which this article attempts to clear up.

In my spare time I've been known to peruse various websites, newsgroups, and other online forums devoted to all things loudspeaker. Over the years I've learned quite a bit, gaining further insights into the art and science of speaker building from both luminary and amateur enthusiasts alike. And if you, like me, spend any time following the variegated tracery of threads which make up these various online resources, you'll also know the noise-to-useful-information ratio can be very large indeed.

I've often seen it happen that an idea or concept, lacking little or no technical merit or validity, is promulgated as truth. I've elevated to the rank of "myth" those seemingly inevitable items that pop up year after year. I've looked at some of my favorites under the objective light of hard science, and I'd like to share the outcome of my own investigation into these favorite myths of mine.

## 1. DAMPING FACTOR

Quite a large amount of online text has been devoted to this item in the forums I mentioned earlier. Reading the various discussions raging back and forth between contributors over the years, I get the general sense that most folks make too much of the amplifier's contribution to the total system damping factor—and its effect on system performance.

Damping factor (DF) is defined as the ratio of the load impedance (usually taken as 8Ω) to source impedance. It is a dimensionless figure of merit and therefore has no units ascribed to it. In practical terms, when applied to an amplifier, it indicates to what degree

the amp can dissipate the energy generated by a driver's voice coil as the coil's post-impulse motion decays to zero: thus is the mechanical vibration electrically damped.

But focusing solely on DF as determined by an amplifier's output resistance ( $R_g$ ) is simply too narrow in scope; its  $R_{source}$ , the sum of  $R_g$ , cable resistance ( $R_{cable}$ ), crossover network component resistance ( $R_{component}$ ), and  $R_{vc}$  (voice coil resistance) determine the true DF as seen by the driver. In fact, focusing on  $R_g$  alone also ignores the single largest contributor to DF: the driver's voice coil.

To resolve a clearer picture of DF, I ran a series of simulations to determine to what degree various damping factor values as determined by  $R_{vc}$ ,  $R_g$ ,  $R_{cable}$ , and  $R_{component}$  affect system response performance at resonance frequency.

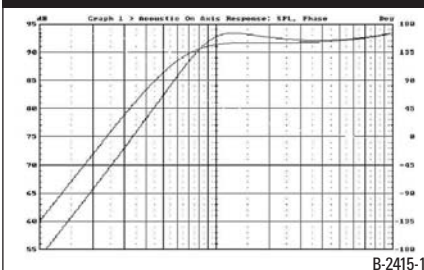
In the first simulation set, I used a test model representing a closed-box system, whose initial design was based upon  $DF = 8$ ,  $R_{source} = 0\Omega$ , where:

$$R_{source} = R_g + R_{cable} + R_{component} (\Omega)$$

and a  $Q_{tc} = 1/\sqrt{2}$ , (Butterworth B2 alignment) producing a maximally flat response, with zero peak at resonance.

Although DF is typically specified at a load impedance value of 8Ω, I used  $R_{vc} = 6.2\Omega$ , which was the  $R_{vc}$  parameter value of the driver I modeled (KEF B139). If you're more comfortable thinking of DF in terms of the usual 8Ω load, convert the given 6.2Ω-based  $R_g$  figure by multiplying the given val-

FIGURE 1: System response at  $DF = 1$  and  $DF = \infty$ .



ues by 1.2903. At any rate, in running the analysis, some interesting trends emerged from the data generated by the simulations (*Fig. 1*).

As I mentioned earlier, the  $R_g$ , or amplifier output resistance, in actual practice is only part of  $R_{source}$ ; all three components of  $R_{source}$ , along with  $R_{vc}$ , combine to affect DF. Of all resistance component values, however,  $R_{vc}$  is typically by far the largest and therefore has the greatest effect on DF.

$R_{source}$  affects  $Q_{ec}$ , which in turn affects system QTC. Rewriting the well-known formula for calculating QTC from  $Q_{ec}$  and  $Q_{mc}$ , but taking into account  $R_g$ ,  $R_{vc}$ ,  $R_{cable}$ , and  $R_{component}$ , gives:

$$Q_{ec}' = 2\pi * f_c * M_{mc} * \frac{R_{vc} + R_{source}}{B^2 l^2}$$

where:  $Q_{ec}'$  = system Q at resonance ( $f_c$ ), due to electrical losses, including  $R_{source}$ , dimensionless

$f_c$  = system resonance of totally enclosed box (TEB), Hz

$B l$  = force factor, magnetic flux density in voice coil gap \* coil winding length, TM

$$R_{source} = R_{cable} + R_{component} + R_g (\Omega)$$

Plug  $Q_{ec}'$  into the well-known equation used to calculate  $Q_{tc}$ .

$$Q_{tc}' = \frac{Q_{ec}' * Q_{mc}}{Q_{ec}' + Q_{mc}}$$

where:

$Q_{tc}'$  = total system Q at resonance ( $f_c$ )

due to both mechanical and electrical losses, taking into account  $R_{source}$ , dimensionless

$Q_{mc}$  = system  $Q$  at resonance ( $f_c$ ), due to mechanical losses, dimensionless

A cursory examination of these formulae shows that as  $R_g$  increases, so too does  $Q_{tc}$ , and with the increase in  $Q_{tc}$  comes an increasing response peak at resonance. And, of course, adding the other two components that make up  $R_{source}$ ,  $R_{cable}$ , and  $R_{component}$  produces even larger changes in  $Q_{ec}$ ,  $Q_{tc}$ , and system response at resonance (Figs. 2-5).

The question is, at what point does the effect of declining  $DF$  become audible? Looking at the first dataset, which I generated considering  $R_{source}$ , it would appear that only in the very worst cases do these changes enter the realm of possibly audible. And setting the upper value limit of  $R_{source} = R_{vc}$  is certainly an extreme case (Table 1). Noting the position of  $R_{source}$  in the  $Q_{ec}$  equation tells you that keeping it as low as possible is helpful, but only modestly so given that  $R_{vc}$ , under typical circumstances, is much, much larger.

The key to understanding the true nature of  $DF$  is realizing that it's the *total resistance seen by the driver's motor coil* that counts and that  $R_{vc}$  is—by

far—the largest, single most important contributor to the total resistance value of the amp/cable/component/voice-coil circuit. Because the value of  $R_{source}$  tends to be  $\ll R_{vc}$ ,  $R_{source}$  doesn't much change the degree of damping the driver experiences at resonance owing to its own  $R_{vc}$ . Why?

Damping is energy dissipation—here in the form of direct conversion of electrical power into heat, with the end result said energy is no longer stored in the system. Typically, 95%+ of the power supplied by the amplifier winds up as heat generated by the voice coil.

To get a clearer picture of where and to what degree dissipation occurs within the amp/cable/component/voice-coil circuit, I ran a simulation of a driver based on a circuit (Fig. 6), the schematic of which some of you may recognize in my earlier *Speaker Builder* article ("Pspice Probes a Loudspeaker's Evolution," 1/99, p. 26). In that article, I demonstrated how to produce an accurate model (with both temperature- and frequency-dependent components) of a raw driver using PSpice. I've added  $R_g$ ,  $R_{cable}$ , and  $R_{component}$  values, retaining the original model  $R_{vc}$  value of  $4.15\Omega$ , found in that earlier article. I then probed for power dissipation at each of the four components to see how and where power was being dissipated. If you'd like to experiment with the model using your own circuit editor/analyzer,

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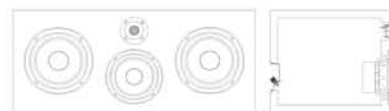
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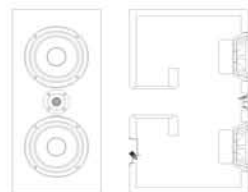


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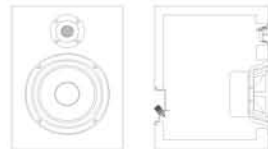


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**FIGURE 2: Attenuation (dB) vs. damping factor.**

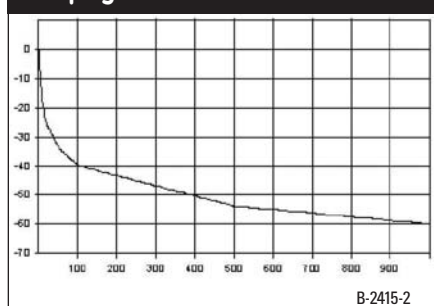


**FIGURE 3: Resonance peak (dB) vs. damping factor**

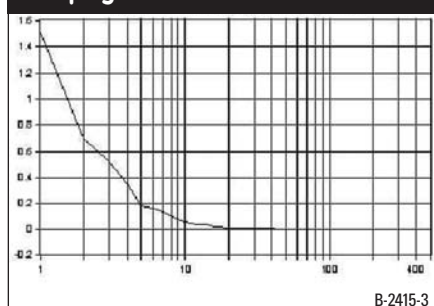


**FIGURE 4: QTC vs. Rg (ohms).**

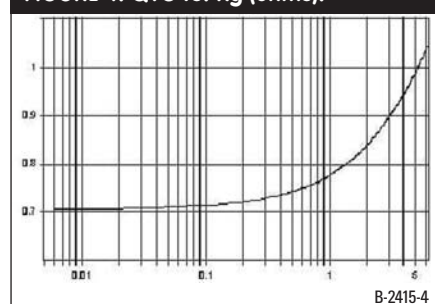


**FIGURE 5: QTC vs. damping factor.**

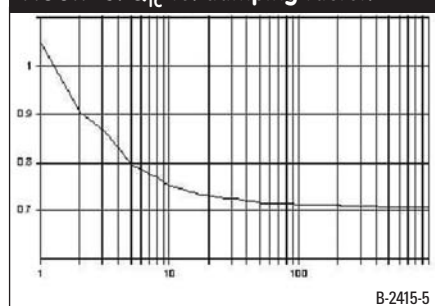


**TABLE 1: DF, Rsource, and resulting variations in electrical attenuation, resonance peak, and Qtc'**

| DF     | Elec. Attn.<br>(dB) | Rsource<br>(Ω) | Resonance<br>Peak (dB) | Qtc'   |
|--------|---------------------|----------------|------------------------|--------|
| ∞      | -20*Log(∞)          | 0.0000         | 0.00000                | 0.7071 |
| 1000.0 | -60.000             | 0.0062         | 0.00000                | 0.7075 |
| 500.0  | -53.970             | 0.0124         | 0.00000                | 0.7079 |
| 100.0  | -40.000             | 0.0620         | 0.00060                | 0.7116 |
| 50.0   | -33.979             | 0.1240         | 0.00270                | 0.7161 |
| 20.0   | -26.020             | 0.3100         | 0.01580                | 0.7295 |
| 15.0   | -23.520             | 0.4133         | 0.02717                | 0.7368 |
| 10.0   | -20.000             | 0.6200         | 0.05676                | 0.7512 |
| 5.0    | -13.979             | 1.2400         | 0.18422                | 0.7925 |
| 2.0    | -6.020              | 3.1000         | 0.69477                | 0.9013 |
| 1.0    | 0.000               | 6.2000         | 1.50960                | 1.0447 |

**FIGURE 7: SPICE net list for test circuit.**

```

CIRCUIT DESCRIPTION ****
*****
.PARAM      Temperature=20
** Analysis setup **
.ac DEC 101 10 40K
.STEP PARAM Temperature LIST
+ 20 80 160 250
** Schematics Netlist **
R_Dumb      0 SN_0001 99999E99999
R_Dumber    0 SN_0002 99999E99999
G_Lem       SN_0001 SN_0002 LAPLACE { V(SN_0001, SN_0002) }
           { + 1/(s*.0324076*pwr(abs(s),(-.4202))) }
C_Cmes      0 SN_0002 857.31796E-6
R_Res       0 SN_0002 19.700547
L_Lces      SN_0002 0 31.400591mH
G_Rem       SN_0003 SN_0001 LAPLACE { V(SN_0003, SN_0001) }
           { + 1/(.0036483*pwr(abs(s),(.7358))) }
V_Vac       SN_0004 0 DC 0V AC 2.828V
R_Rg        SN_0004 SN_0005 .016
R_Rcable    SN_0005 SN_0006 .15
R_Rcomponent SN_0006 SN_0007 .3
R_RevC      SN_0007 SN_0003 {4.1500*(1+.00393*(Temperature-20))}

```

I've included the netlist generated by PSpice (Fig. 7).

Once I ran the simulations it was easy to see from the generated output graphs that, indeed, the majority of dissipation—and therefore damping—was taking place in the RevC component of the circuit, i.e., the driver's voice coil

(Fig. 8). In the graph, the topmost trace is that of RevC, which clearly indicates that RevC is the point in the circuit where the most power is being dissipated.

Hence, it has the greatest degree of control in determining the DF of the circuit. By this point in the analysis

dB SPL loss, separate from DF, where:

$$\text{dB SPL loss} = 20 * \log\left(\frac{Z_{\text{vc}}}{Z_{\text{vc}} + R_{\text{source}}}\right)$$

Here, Zvc represents the frequency-dependent impedance of the driver's voice coil, in ohms. This equation illustrates that increasing Rsource results in greater dB SPL losses. Given this fact, best practices dictate keeping Rsource as low as possible.

From the preceding investigation I think it fair to draw the conclusion that amplifier DF has very, very little effect on the total circuit DF, and what little effect it does have occurs in only the very worst of cases.

## 2. MAGNETIC SHIELDING

Online threads typically start with someone bemoaning the fact that their latest design, now made reality, can't be placed next to their TV or CRT computer monitor without causing the display

to become distorted. What to do?

Soon there follows lots of well-intentioned suggestions—frequently possessing scant traces of technical merit—to shield or enclose the monitor or loudspeaker with a variety of metals. To sort the rubbish from the useful suggestions, I ran a series of Finite Element Analysis (FEA) analyses, each modeling a different suggestion on how to conquer video display distortion. My analyses

**FIGURE 6: Test circuit.**

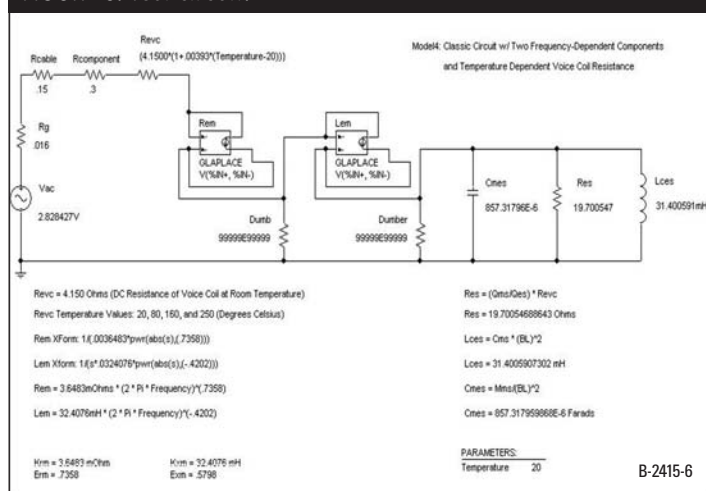


**TABLE 2: Metal Properties**

|               | Resistivity<br>(Ω · cm) | Density<br>(gm · cc) | Mag. Susc.<br>(cgs/g)                          | Temperature<br>Coefficient | UNS ID<br>Number | Notes                                             |
|---------------|-------------------------|----------------------|------------------------------------------------|----------------------------|------------------|---------------------------------------------------|
| Silver (AG)   | 1.55e-006               | 10.491               | 2e-007                                         | 0.0038                     | -                | Pure, NF                                          |
| Copper (Cu)   | 1.70e-006               | 8.96                 | -8e-008                                        | 0.00382                    | -                | Cold Drawn, Pure, NF                              |
| Copper (Cu)   | 1.71e-006               | 8.89-8.94            | -1.04 × 10 <sup>-6</sup><br>cm <sup>3</sup> /g | 0.00382                    | C10100           | OFC, NF                                           |
| Gold (Au)     | 2.22e-006               | 19.32                | -1.42e-007                                     | 0.0034                     | -                | Noble Metal: does not<br>react with air. Pure, NF |
| Aluminum (Al) | 2.7e-006                | 2.6989               | 6e-007                                         | 0.0039                     | -                | Pure, NF                                          |

NF = Non-ferromagnetic

will show, in the end, the old saw "an ounce of prevention is worth a pound of cure" definitely holds true where it comes to effectively corralling stray magnetic fields.

The first simulation is that of a magnet/steel structure with its magnetic field free to fill the air surrounding it (Fig. 9). The squiggly lines surrounding the structure are magnetic field lines, scaled in webers. Any electrons streaming through the magnetic field (such as those found in an energized cathode ray tube) will be pulled off course, resulting in all sorts of image distortions.

One solution, of course, is to keep that magnetic field from getting near the electron path. Moving the field source a sufficient distance from the CRT gun would clear things up, as the farther the distance from the source the lower the flux density, but that only addresses the symptoms, not the underlying pathology. As well, space constraints may make putting enough space between monitor and speaker impractical.

For my second simulation (Fig. 10), I modeled another common suggestion: place a shield between the loudspeaker and the CRT. In this case I placed a sheet of aluminum squarely in the path of the field and it did . . . just about nothing. Look at those lines—they're having a field day (pun intended)!

This looks like a sure-fire formula for disappointment, so obviously it isn't a useful solution. The problem here is that aluminum is a non-ferromagnetic material with a relative permeability value nearly identical to that of air—thus it's about as useful as air in corralling the stray magnetic field and preventing CRT image distortion. Display distortion continues virtually unabated.

For my third simulation (Fig. 11), I modeled another common suggestion:

enclose the loudspeaker in a container of some sort. In this case I used copper as the material. Once again you see the field is anything but corralled, which is just what you would expect using yet another non-ferromagnetic material.

Copper has a relative permeability very close in value to that of aluminum, and the results from this FEA run reflect this. Any CRT caught within the field surrounding the copper box would suffer distortions just as it would with the aluminum shield modeled previously.

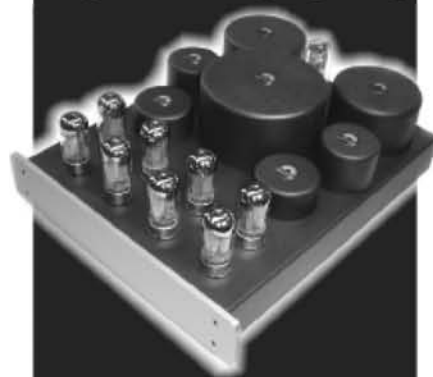
You could reasonably argue that other materials, such as steel, mumetal, and so on, could have been used to produce the modeled enclosure. But their effectiveness, practicality, predictability of results, and so on, when measured against cost and so forth still don't make their use worthwhile.

Besides, suppose you did find a material that could effectively shield the CRT monitor from the magnetic field surrounding the loudspeaker system. You'd need to encase the entire system in order to maximize effectiveness. And who wants to listen to a loudspeaker entirely contained within a metal box? Not very practical.

In my fourth FEA simulation (Fig. 12) I modeled the magnet structure with a second magnet (also known as a bucking magnet) of the correct size, shape, material, and orientation placed up against the original magnet structure. Here you can see that the magnetic field is very effectively corralled, certainly far better than in any of the preceding simulations. Best practices here dictate that using drivers with a built-in bucking magnet already in place is the correct way to prevent CRT-distorting magnetic fields getting at your TV or computer monitor.

The bonus? The presence of the buck-

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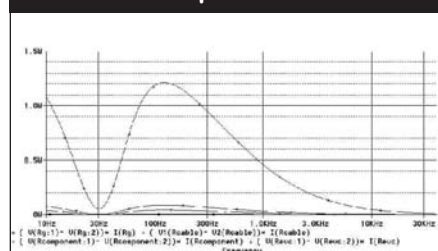
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**or 2 x 90CT 11A**  
**or 90CT 22A**  
**or 180CT 11A**  
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Finish **Textured black**  
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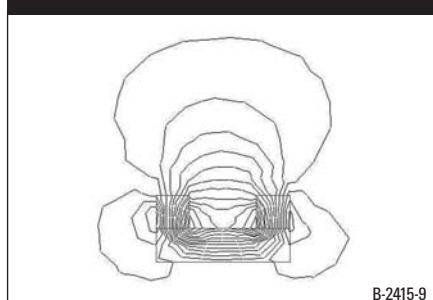
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**FIGURE 8: Power dissipation among the Rsource components.**



B-2415-8

**FIGURE 9: Unrestrained magnetic field.**



B-2415-9

**TABLE 3: Gauge wire dimensions**

|               | Length  | Diameter<br>(inches) | Area<br>(cir mills) | Area<br>(in <sup>2</sup> ) |
|---------------|---------|----------------------|---------------------|----------------------------|
| 30-gauge wire | 57.619' | 0.01003              | 100.5               | 0.00007894                 |

ing magnet can increase field density in the voice-coil gap, which, in turn, means increases in efficiency and lowered QES, as compared to its non-bucking equipped twin. Conversely, placing a cup shield made out of a ferromagnetic material around the magnet assembly will typically result in a reduction in field gap density, which results in reduced Bl, efficiency, and increased Q<sub>ES</sub>—all because the cup provided a lower reluctance path for the flux to follow than that within the voice coil gap.

### 3. GOLD IS THE BEST CONDUCTOR

Often found buried in online discussions of cables and interconnects is the contention that gold is the best conductor from which to make said items. Reality is, whether or not gold (or any other metal, for that matter) is the best choice of conductor depends on the application for which it is intended.



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I've put together a chart (*Table 2*) listing basic properties of several metals I've seen mentioned at one time or another in online discussions. From a material properties point of view, gold's most outstanding feature is that it's a noble metal and therefore does not react with the environment as the other metals do, which makes it a great choice where corrosion-resistant connectors are required.

A quick look at the chart shows it is not the least resistive or lightest of metals either—two reasons you don't see gold voice coils, especially when you consider that copper is less resistive, lighter, cheaper, and commonly available in wire form. To put the various metals into perspective-enhancing rankings, I constructed a chart which shows a variety of parameters for a theoretical voice coil, consisting of a 4-layer coil, wound with 30-gauge wire on a 2" former.

Using:

$$R = \rho * l / A \text{ } (\Omega)$$

**FIGURE 10: Magnetic field with aluminum shield.**

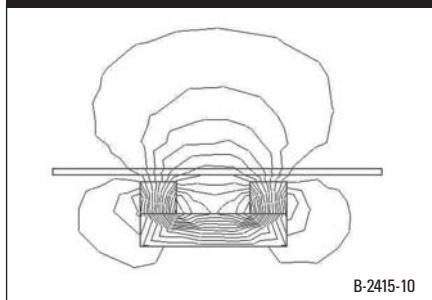
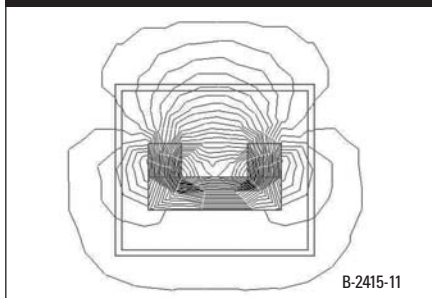


**FIGURE 11: Magnetic field with copper box.**



where: R = resistance of sample at 20°C  
ρ = specific resistance  
l = length of sample  
A = cross-sectional area of sample

to determine resistance of the conductor and using:

$$R = R_{ref} * [1 + a * (T - T_{ref})] \text{ } (\Omega)$$

where: R = conductor resistance at temperature "T."

R<sub>ref</sub> = conductor resistance at 20°C

a = temperature coefficient of resistance

T<sub>ref</sub> = reference temperature at which R<sub>ref</sub> is specified.

to determine resistance at various temperatures. And in each case, as I mentioned previously, 30-gauge wire with the dimensional properties listed in *Table 3* is used.

From *Table 4*, you can easily see that a voice coil made of either silver, copper, or OFC copper has a lower R<sub>evc</sub> than one made of gold. You can also see that these voice coils would be lighter than one made of gold. It is interesting to note that an OFC copper voice coil has a slightly *higher* resistance than the non-OFC version.

Silver, though the best conductor in

**FIGURE 12: Magnetic field with bucking magnet.**

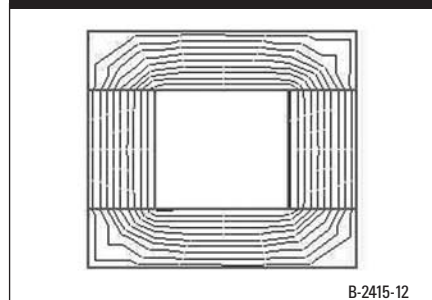
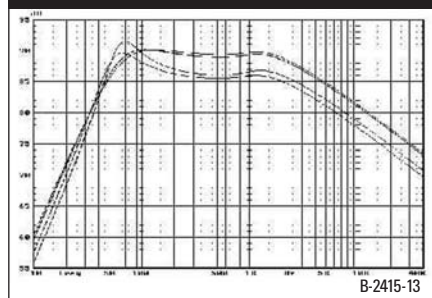


**FIGURE 13: System response resulting from voice coils made of various metals.**



terms of exhibiting the lowest resistance, isn't often used as voice-coil wire because (aside from cost) the gain in driver efficiency as a consequence of lowered resistivity is more than offset by the loss of efficiency caused by the increase in voice-coil mass. Conversely, aluminum does find its use in tweeters because its comparatively lesser density makes for a lower moving mass voice coil, which offsets the decreased efficiency owing to its higher resistivity.

I was particularly interested in seeing how the weight differences between voice coils made of the differing materials would affect system dB SPL response and impedance, so I ran five simulations in LEAP. In each case I held Revc constant, varying BL, Mmd (diaphragm mass, in grams), and letting LEAP alter the remaining driver parameters. I then (virtually) placed them—one at a time—in the same enclosure and generated db SPL and impedance plots.

It's interesting to note in the dB SPL graph (Fig. 13) that when Revc was held constant, the drivers sporting voice coils made from aluminum and gold (the lightest and heaviest materials modeled) had a response ~3db lower than those systems modeled with either version of copper or silver. Silver produced the highest reference efficiency system, but only a small fraction of a dB above either of the coppers.

In looking at the impedance chart, you see the two coppers present nearly identical curves (Fig. 14). Aluminum had the lowest Zpeak (~31Ω) of all samples and silver had the highest (~81Ω). Gold presented an impedance peak located between that of aluminum and the two coppers.

When you take into consideration other issues important in the manufacturing process (cost, availability,

**TABLE 4: Various wire materials and their effects on resistance, attenuation, and voice coil weight**

|                 | Resistance (Ω) at 20°C | Attenuation Loss* at 20°C (dB) | Resistance (Ω) at 100°C | Attenuation Loss* at 100°C (dB) | Weight (gm)  |
|-----------------|------------------------|--------------------------------|-------------------------|---------------------------------|--------------|
| Silver (Ag)     | 5.345                  | -4.44                          | 6.97                    | -5.443                          | 9.383        |
| Copper (Cu)     | 5.862                  | -4.774                         | 7.653                   | -5.830                          | 8.01         |
| OFC Copper (Cu) | 5.896                  | -4.796                         | 7.815                   | -5.919                          | 7.95 - 7.996 |
| Gold (Au)       | 7.655                  | -5.8312                        | 9.737                   | -6.915                          | 17.280       |
| Aluminum (Al)   | 9.31                   | -6.704                         | 12.215                  | -8.052                          | 2.413        |

\*Referenced to 8Ω

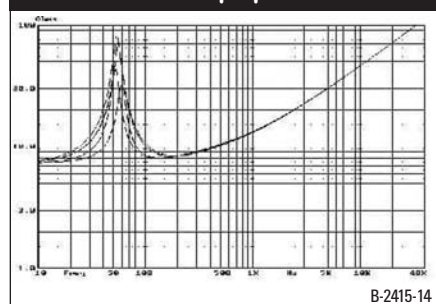
resistance to mechanical and thermal stress, solderability, and so on) and add them in to the equation with copper's comparatively low resistance and light weight, it's easy to see why it features so prominently in audio. Given the preceding analysis, I think it's fair to say that for applications other than those requiring a high degree of corrosion resistance, gold is not the best conductor around, and in most other cases good ol' copper (OFC or not) is the best choice.

#### 4. BASS REFLEX IS MORE EFFICIENT THAN TOTALLY ENCLOSED BOX

Here's one that's been around far longer than any Internet online forum! To sort the facts from fiction, I ran five systems simulations in LEAP, using a driver with the specs listed in Table 5.

In the first simulation, I modeled a totally enclosed box system, used as a baseline against which all subsequent modeled systems were compared. I then

**FIGURE 14: Metals properties table.**



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**TABLE 5: Test Driver Specs**

|                       |                               |                             |
|-----------------------|-------------------------------|-----------------------------|
| SPLo=89.93dB          | SPLi = 89.29dB                | SPLi = 89.82dBI             |
| no = 0.62%            | $\eta_l = 0.53\%$             | Eg = 2.83V                  |
| Rem = 3.56 $\Omega$   | Lem = 1.264mH @ 1kHz          |                             |
| Rem = 34.26 $\Omega$  | Lem = 0.457mH @ 20kHz         |                             |
| Znom = 8.000 $\Omega$ | Sd = 0.0349m <sup>2</sup>     | Fi = 30.218Hz               |
| Revc = 7.093 $\Omega$ | BL = 12.7619 T • m            | Fo = 31.328Hz               |
| Krm = 4.779m $\Omega$ | V <sub>AS</sub> = 89.0391 ltr | Q <sub>MS</sub> = 6.054     |
| Kxm = 24.731mH        | Cms = 514.7923 $\mu$ M/N      | Q <sub>ES</sub> = 0.430     |
| Erm = 0.756           | Mms = 50.1364gram             | Q <sub>TS</sub> = 0.401     |
| Exm = 0.660           | Mmd = 46.3875gram             | Pmx = 100.000               |
|                       |                               | Wms $\Theta_{vc}$ = 2500C/W |

ran three subsequent simulations, each representing some permutation of the first baseline system. I then ran a fifth and final simulation, modeling a properly designed and tuned BR system. The modeled systems' specs are shown in Table 6.

I started with a comparison of the

mind that each vertical dB SPL axis is scaled in 1dB units).

I then re-ran the simulation, this time modeling a system with a net internal cabinet volume ten times larger than that of the baseline system (Fig. 16). Even with a one order of magnitude increase in net Vb, I observed no change

baseline system to a system in which Vb was doubled, with all other design parameters held constant. This doubling of the net internal cabinet volume used for the baseline design did very little to change overall response and nothing in the passband reference efficiency region as shown in Fig. 15. (As you read through the graph data, keep in

in  $\eta_{odB}$  of the system; it still hovers extremely close to that of the driver.

Next, I took the baseline TEB system and virtually cut a port in the cabinet. The port modeled had a 0.75" depth, and a cross-sectional area of  $0.5 \cdot S_d$  (0.17545m<sup>2</sup>). All other parameters were, as before, held constant. I then re-ran the simulation, producing the result shown in Fig. 17.

There are some noteworthy items in this graph. First, you see the mid-band reference efficiency of the ported system has actually *dropped*, while at the same time a ~+10dB local maximum has appeared at just a bit above 100Hz.

I've included beneath each graph an area under curve (AUC) value generated by performing an integration on the response plot from 10Hz to 40kHz. The  $|\Delta|$  value represents the absolute value of the percentage difference found when comparing the AUC value of the baseline TEB versus each of the other four systems modeled. I included the

**FIGURE 15: Response plots for baseline TSB and 2x VB TEB. Baseline AUC: 3138748 2x TEB AUC: 3138695  $|\Delta| = .001\%$ .**



**FIGURE 16: Response plots for baseline TEB and 10x VB TEB. Baseline AUC: 3138748 10x TEB AUC: 3138627  $|\Delta| = .0038\%$ .**

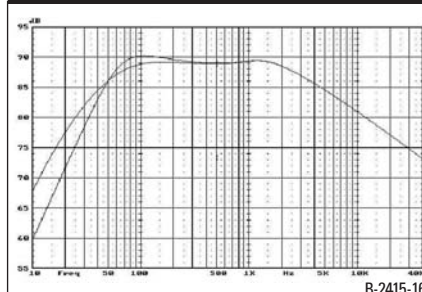
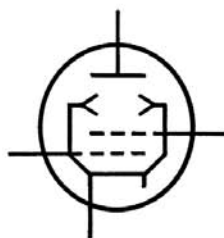
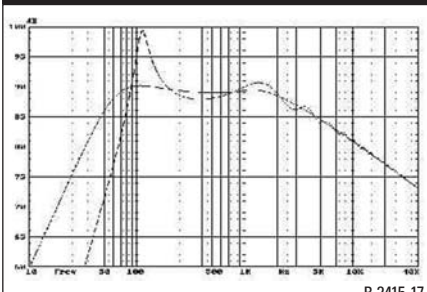


**FIGURE 17: Response plots for baseline TEB and ported TEB. Baseline AUC: 3138748 Ported TEB AUC: 3138952  $|\Delta| = .006\%$ .**



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AUC as had there been any substantial differences in the resulting simulation  $\eta_{0dB}$  values, it would have shown up as a  $|\Delta|$  far larger than the vanishingly small values shown.

Last, I compared the baseline TEB response with that of a BR system, featuring, as in each preceding case, the test driver specified above (Fig. 18). Here you observe the bandwidth extension which you would expect for a properly designed and tuned BR cabinet. Once again, there is no change evident in the observed passband  $\eta_{0dB}$  value.

From these graphs you can conclude that simply cutting a hole in a TEB cabinet or increasing the size of the cabinet will not increase  $\eta_{0dB}$  values. You've also seen that, in some cases, such modifications can slightly decrease system  $\eta_{0dB}$ .

You've also observed you can design a TEB and BR system with the same driver and come up with a BR system that sports an extended LF response when compared with that of the TEB, but that won't change the  $\eta_{0dB}$  value observed in either system, as the graphs clearly indicate. An increase in bandwidth—yes, but a change in reference efficiency—no.

To understand why, for each of the systems modeled, the  $\eta_{0dB}$  values determinedly remained within a small fraction of a dB of the test driver's reference efficiency value, take a look at the mathematical underpinnings of driver and system efficiency. You'll also see why system  $\eta_{0dB}$  hovers near or below driver  $\eta_{0dB}$  but never exceeds the driver's  $\eta_{0dB}$  value.

From:

$$\eta_0 = \frac{\rho_0}{2\pi c} * \frac{B^2 l^2}{Revc} * \frac{Sd^2}{Mms^2}$$

you can calculate a driver's reference efficiency, where:

$\eta_0$  = reference efficiency of the driver, dimensionless

$\rho_0$  = the density of air at STP, 1.204kg/m<sup>3</sup> at 20°C

$c$  = speed of sound at STP, 343.4m/s at 20°C

$B$  = magnetic flux density in the driver air gap, in teslas

$l$  = length of the voice coil winding in the gap, m

$Revc$  = DC resistance of the driver

voice coil, ohms

$Mms$  = mechanical mass of the driver diaphragm assembly, including air load, kg

$Sd$  = effective projected surface area of the driver diaphragm, m<sup>2</sup>

From:

$$\eta_0 = k_\eta * f_3^3 * Vb$$

you can calculate the reference efficiency of a closed box system where:

$\eta_0$  = reference efficiency of the system, dimensionless

$k_\eta$  = efficiency constant, dependent upon chosen alignment, s<sup>3</sup>m<sup>-3</sup>, given

by:

$$k_\eta = \frac{4\pi^2}{c^3} * \frac{fc^3}{f_3^3} * \frac{Vat}{Vb} * \frac{1}{Qec}$$

$fc$  = closed box system resonant

**FIGURE 18: Response plots for baseline TEB and BR system. Baseline AUC: 3138748 BRLEF AUC: 3139468  $|\Delta| = .0229\%$ .**

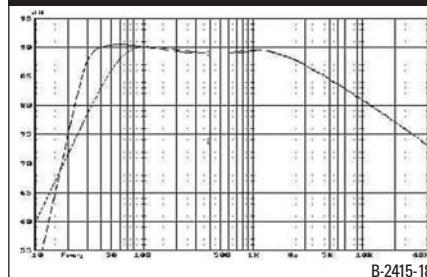


**TABLE 6: Observed  $\eta_{0dB}$  (@ 500Hz),  $Vb$  (ft<sup>3</sup>) for the various systems modeled**

|              | $Qb$ | $Vb$ (ft <sup>3</sup> ) | Observed $\eta_{0dB}$<br>(at 500Hz) | Notes              |
|--------------|------|-------------------------|-------------------------------------|--------------------|
| Test Driver  | -    | -                       | 89.29dB                             | Raw driver         |
| Baseline TEB | 2    | 0.9046                  | 89.29dB                             | 100% stuffed       |
| 2× VB TEB    | 2    | 1.809                   | 88.94dB                             | Baseline TEB VB*2  |
| 10× TEB      | 2    | 18.09                   | 88.87                               | Baseline TEB VB*10 |
| Vented TEB   | 2    | 0.9046                  | 87.91dB                             | Port area = 0.5Sd  |
| BR system    | 7    | 2.154                   | 88.95dB                             | $f_b = 31.55Hz$    |

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frequency

$f_3$  = -3dB cutoff frequency of the system

$c$  = speed of sound at STP, 343.4m/s at 20°C

$Q_{ec}$  = system electrical Q at  $f_c$

$V_b$  = net internal volume of enclosure

$V_{at}$  = total system compliance expressed as equivalent volume of air, l

For closed box systems,  $k_\eta$  maxes out at  $\sim 2.0 \cdot 10^{-6}$ . This figure represents a maximum performance limit and can be reached only by employing a lossless system, based on the C2 alignment.

For vented systems you have, once again:

$$\eta_0 = k_\eta \cdot f_3^3 \cdot V_b$$

where:

$\eta_0$  = reference efficiency of the system, dimensionless

$k_\eta$  = efficiency constant, dependent upon chosen alignment,  $s^3m^{-3}$ , given by:

$$k_\eta = \frac{4\pi^2}{c^3} \cdot \frac{f_s^3}{f_3^3} \cdot \frac{V_{as}}{V_b} \cdot \frac{1}{Q_{es}}$$

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$f_s$  = driver resonant frequency

$f_3$  = -3dB cutoff frequency of the system

$c$  = speed of sound at STP, 343.4m/s at 20°C

$Q_{es}$  = driver electrical Q at  $f_s$

$V_{as}$  = volume of air having same acoustic compliance as driver suspension

$V_b$  = net internal volume of enclosure

For vented systems,  $k_\eta$  maxes out at  $\sim 3.9 \cdot 10^{-6}$ . This figure represents a maximum performance limit and can be attained only by employing a lossless vented system, based on a  $k = 0.5$ , C4 alignment.

In all three cases, you can convert  $\eta_0$  values to power sensitivity in decibels by:

$$\eta_{0dB} = 112.2 + 10 \cdot \text{Log}(\eta_0) \\ (\text{dB SPL}/1W/1m)$$

or, alternately, converted to percent by:

$$\eta_{0\%} = 100 \cdot \eta_0 \quad (\%)$$

Efficiency, broadly defined in this context, is the ratio of acoustic power out to electric power in:  $\eta = \frac{P_A}{P_E}$

where both  $P_A$  and  $P_E$  are in watts.

The two system reference efficiency equations say that enclosure volume, cutoff frequency, and maximum reference efficiency are interdependent. Small refers to this as the "Efficiency-Bandwidth-Volume Exchange." Given the alignment type and losses present,  $\eta_0$  can be maintained while altering either  $f_3$  or  $V_b$  only by altering the value of the remaining factor. Want a lower  $f_3$ ? You'll need to increase  $V_b$ . Want a smaller cabinet? You can expect  $f_3$  to rise, presuming, of course, that you want to maintain a particular  $\eta_0$  value constant (or ratio of acoustic power out to electric power in).

Keep in mind that the calculated system  $\eta_0$  values represent an ideal performance maximum: the best you could hope for, given a specific alignment resulting from a specific design for a specific driver. In practice, actual system  $\eta_0$  values are lowered or constrained by such factors as driver mechanical losses, enclosure losses, and employ-

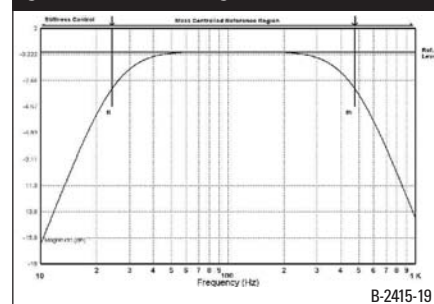
ing alignments other than those mentioned previously as being maximally efficient. Therefore, it's not difficult—by choice or chance—to design a system that exhibits a lesser actual  $\eta_0$  value.

Loudspeaker design is a study in artful compromises; what you trade away in one area to gain in another depends upon, among other things, what your design priorities and goals are. If, in this case, for a given driver every design permutation you model delivers insufficient  $\eta_{0dB}$  values, the only way to increase the values would be to switch to a more efficient driver.

Above resonance, but below  $f_h$ , the driver is operating in its inertial mass-controlled reference efficiency pass-band. As such, in this portion of its output spectrum, the cone is mass-limited and the driver functions as a constant acceleration device, producing constant volume velocity. An enclosure does not change the mass of the driver cone, so it cannot affect efficiency as constrained/controlled by cone mass. The air within the cabinet does not provide resistance to the motion of the cone; if it did, varying cabinet size would produce varying observed  $\eta_{0dB}$  levels. You do not observe such variations across the five simulations (even in the case where cabinet size is increased by a factor of 10).

Given that a driver's reference efficiency is taken from the mass-controlled region of its response, a driver's efficiency is independent of an enclosure's volume, whether or not said enclosure is vented. Neither does an enclosure alter a driver's  $Bl$ ,  $S_d$ , or  $M_{ms}$ . So essentially you see no real variations in  $\eta_{0dB}$  levels across each of the five simulations because the cabinet does not affect or otherwise alter the driver parameters that determine its reference

**FIGURE 19: Bandpass response of a generic electromagnetic driver.**



efficiency. Hence you never see a system sporting  $\eta_{\text{odB}}$  levels greater than that of the driver, but given a variety of factors you can see a cabinet/driver combination that lowers  $\eta_{\text{odB}}$  levels that could otherwise be attained.

## 5. THE PERFECT DRIVER. . .

would be infinitely stiff and completely massless.

I think this myth has survived as long as it has because, at first glance, the assertion seems to make sense; after all, a driver that is infinitely stiff and has no mass should have no problem responding perfectly and instantly to every electrical impulse fed its inputs, right? Let me show you what a bit of examination behind the scenes of the typical bandpass response of moving coil driver turns up (Fig. 19).

You can divide the response curve roughly into three regions, each demarcated by  $f_l$  (lower resonance) and  $f_h$  (upper resonance).

Below  $f_l$ , stiffness (or its inverse, compliance—as supplied by the driver's suspension—expressed as  $C_{ms}$ ) predominates in limiting the motion of the cone and output is proportional to frequency (see Joseph D'Appolito, *Testing Loudspeakers*), which, below resonance, means a rolloff of 12dB/octave as frequency drops. In this region the driver functions as a constant excursion device.

At  $f_l$ , output depends on resistance, including  $R_{\text{source}}$  and  $R_{\text{vc}}$ . The frequency at which  $f_l$  occurs is determined by the moving mass of the driver. And as the theoretical moving mass of the driver  $\rightarrow 0$ ,  $f_l$  approaches  $\infty$  Hz. As shown by:

$$f_s = .5p * (1/(C_{ms} * M_{ms}))^{0.5} \quad (\text{Hz})$$

where:

$f_s$  = free air resonance of driver, Hz

$C_{ms}$  = mechanical compliance of driver, m/N

$M_{ms}$  = effective mechanical mass of driver diaphragm, in kg

Between  $f_l$  and  $f_h$  is the mass-controlled reference region of the driver. In this portion of its output spectrum, the cone is mass-limited and the driver functions as a constant acceleration device, producing constant volume ve-

locity. Radiation remains essentially hemispherical and directivity, constant.

At  $f_h$ , like  $f_l$ , resistance plays the controlling role.  $F_h$  can be determined by:

$$f_h = \frac{\omega \cdot L_{\text{vc}} B^2 I^2}{\omega^2 \cdot L_{\text{vc}}^2 + (R_g + R_{\text{vc}})^2}$$

and total resistance at  $f_h$  can be found by:

$$R_{\text{total}} = \frac{(R_g + R_{\text{vc}}) \cdot B^2 I^2}{(R_g + R_{\text{vc}})^2 + \omega^2 \cdot L_{\text{vc}}^2}^{10}$$

So where's the problem with the infinitely stiff/massless driver?

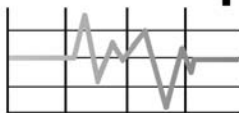
By setting stiffness to  $\infty$ ,  $f_l$  is pushed up to  $\infty$  Hz (besides sounding painful in an abstract sort of way), which means that since response rolls off at 12dB/octave, by the time frequency has dropped to within the audible range response is  $\infty$  dB down, hence no output.

Above  $f_l = \infty$ , the incredible driver is, predictably, functioning as a constant acceleration device generating a flat frequency response, but as it's all hap-

pening at a frequency of  $\infty$  Hz, there's no audible output. The perfect driver remains—as far as human hearing goes—silent. There exists a whole host of other technical issues that would contribute to the driver's zero output (e.g., out-of-phase cancellation), but just focusing on  $C_{ms} = 1/8$  and  $M_{ms} = 0$  are, in and of themselves, sufficient to silence the myth.

Ferretting out the facts behind a myth—or any facet of loudspeaker design, for that matter—can be enjoyable, satisfying, and enlightening. Like a detective story in which every clue is important, you can find yourself developing an almost childlike curiosity when investigating the facts using whichever tool you have at hand: mathematics, CAD, your calculator, or a workbench with a bunch of analog test gear on it. The Internet, while providing forums for the propagation of audio myths, is also a formidable tool in providing many resources useful to sorting facts from fiction. Perhaps the greatest benefit of all such research is that, in the end, you can only gain in understanding. **aX**

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## CORRECTIONS

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

David Ralph

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

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

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

Tim Smith

tsmith15937@msn.com

## ANALYZER REFINEMENTS

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

I also added a 3½-digit digital panel meter. But after some experience I

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

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

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

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

Paul J. Bruinsma  
The Netherlands

### Richard Crawford responds:

I thank Mr. Bruinsma for sharing his experiences and for his recommendations. I would like to reply to his comments on the input potentiometer and its distortion/noise. First, there can be a type of noise (thermal noise) introduced by the input

potentiometer, particularly at midrange settings. Thermal noise will be present in any potentiometer, regardless of quality. The thermal noise should be small enough not to interfere with the 1PPM capabilities, but it can affect the "noise floor" of the instrument.

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

## DRIVER SPECS

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

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

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

shown).

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

Mathew Marulla  
Nashua, N.H.

#### Mark Sanfilippo responds:

Great idea encapsulating the maths in an Excel spreadsheet!

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

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

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

#### SHOWCASE

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

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

Rudy Dick  
Abbotsford, B.C. Canada

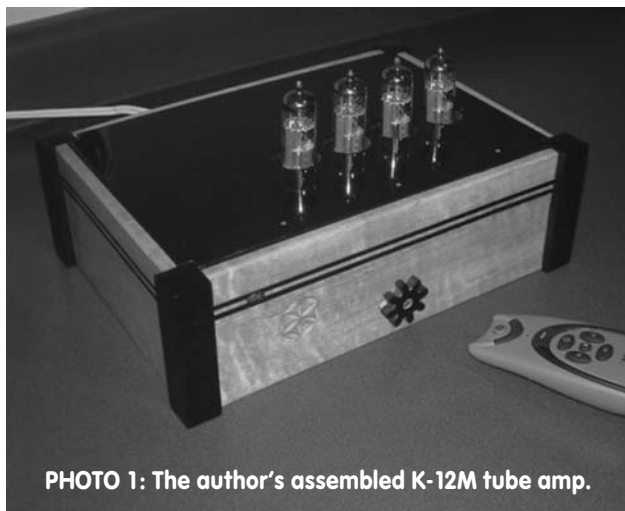


PHOTO 1: The author's assembled K-12M tube amp.

#### SIMPLE AMPS

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

I must say that the articles of Larry Lisle are especially very interesting and correspond very well with my

interests in minimalist amplifier design. With his "Amplification Without Distortion" (July '03 *aX*), "A Minimalist Amplifier Design" (March '05 *aX*), and "A Battery Tube Hifi Amplifier" (GA 3/97) he described very simple amplifiers and gave very good and clear detailed descriptions of the techniques used. His explanations gave insights

into the secrets of these simple amps which are no longer available. Also, his explanation of the filament polarity (in "Amplification Without Distortion") and the (not intended?) negative feedback which results are well worth reading. These details prove that things are not really as they appear at first glimpse. Thank you very much, Mr. Lisle!

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

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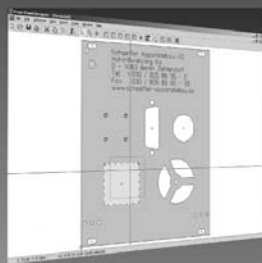
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satisfying answers. Maybe Mr. Lisle can help. In some of his articles the windings of the transformers are additionally loaded with resistors. Also, in a German textbook with similar amplifiers of the 1920s, I have found capacitors across the primary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

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

Ulrich Tönjes  
Bremen, Germany

## Larry Lisle responds:

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

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

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

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

Disadvantages of using "loading" resistors can include increased distortion, loss of gain, and resistor noise. The quality of the transformer, the circuitry, the impedances involved, and the purpose of the

design influence whether or not loading resistors are needed, and their value. For instance, I used a resistor across the secondary of the excellent Hammond 808 transformer in "A Beginner's Push-Pull or Single-Ended Amplifier" (Jan. '02 aX), but not in "A Minimalist Amplifier Design."

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

## SUITABLE OP AMPS

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

David Elderton  
[elderton@telus.net](mailto:elderton@telus.net)

## Gary Galo responds:

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

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

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

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

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

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

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

## HELP WANTED

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

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

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

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

Julian Mitchell  
julian@mitchellstout.co.nz  
New Zealand

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

I have a Dynaco Series II Stereo 70 amplifier. I would like to know how to

make my output tubes EL34/6CA7 last longer.

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

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

Mario Montemurro  
lpolsson@sd52.bc.ca

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

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

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## solid state

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## Three Software Editing Programs

By Gary Galo,  
Regular Contributor

**Sound Forge Audio Studio V. 7.0b.** Sony Media Software, 1617 Sherman Ave., Madison, WI 53704, 608-256-3133; <http://mediasoftware.sonypictures.com>. List price: \$69.95. Street price: \$52.60 from [www.ecost.com](http://www.ecost.com). Requires Windows 2000 or XP.

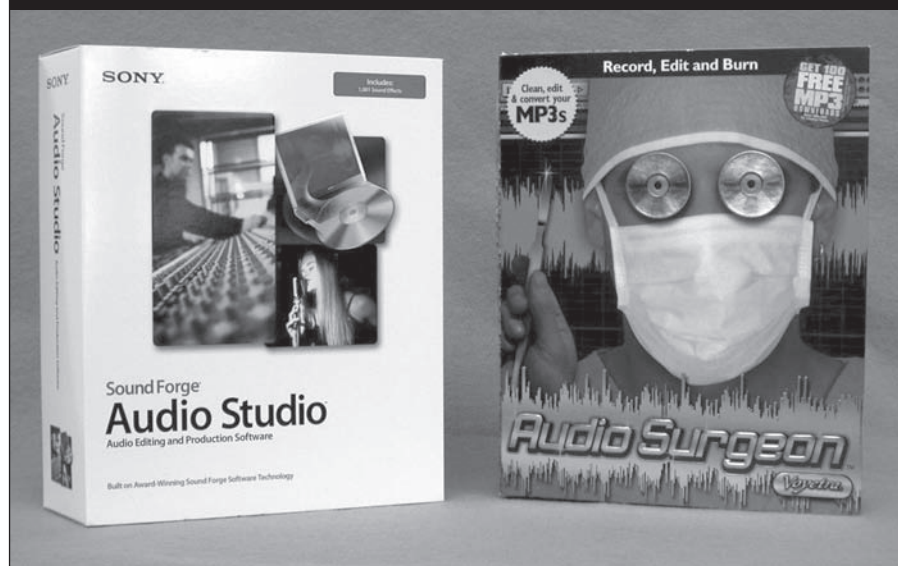
**AudioSurgeon V. 6.** Voyetra Music Software, Voyetra Turtle Beach, 5 Odell Plaza, Yonkers, NY 10701-1406; <http://www.voyetra.com>; 800-233-9377 (sales, in US), 914-966-0600 (sales, non US), 914-966-1102 (Fax); e-mail: [sales@voyetra.com](mailto:sales@voyetra.com); support: [support@voyetra.com](mailto:support@voyetra.com). Download price: \$29.95. Requires Windows 98 or higher.

**DC Six 6.14.** Enhanced Audio, Inc., 3600 Board Road, York, PA 17402, 717-764-9240, 866-260-6376 (toll-free), 717-764-9254 (Fax); [www.enhancedaudio.com](http://www.enhancedaudio.com); [info@enhancedaudio.com](mailto:info@enhancedaudio.com) (e-mail). \$199 (\$99 upgrade from DC Five). Requires Windows 98SE or higher.

For some time I have been using Sony Sound Forge 7.0 digital audio editing software (henceforth referred to as Sound Forge), which I use along with their Noise Reduction 2.0 and CD Architect 5.0 packages. Noise Reduction 2.0 consists of four DirectX audio restoration plug-ins—Vinyl Restoration, Noise Reduction (for hiss

and other continuous noise reduction), Clipped Peak Restoration, and Click and Crackle Removal. All four operate within the Sound Forge editing environment. CD Architect 5.0 is a stand-alone program for Red Book CD mastering with disc-at-once burning. These are full-featured programs intended for professional use.

**PHOTO 1:** Two low-cost digital audio editing programs. Sony Sound Forge Audio Studio offers the speed and user-friendly editing environment of their professional Sound Forge program. Voyetra's AudioSurgeon offers basic editing functions and professional-style disc-at-once CD burning, but CD drive support needs improvement.



## SOUND FORGE AUDIO STUDIO

Sound Forge Audio Studio 7.0b (henceforth referred to as Audio Studio) is Sony's newest consumer version of Sound Forge, replacing Screenblast Sound Forge 7.0 (*Photo 1*). Audio Studio offers many of the features of Sound Forge at a fraction of the price. You can purchase the program either packaged or as a download from the Sony Media Software website. You can also download a fully-functional 30-day trial version.

The Audio Studio interface looks and operates exactly like the professional Sound Forge (*Photo 2*). One of the nicest features of Sound Forge is its lightning-fast speed. On an up-to-date computer you can copy an entire CD worth of data—80 minutes—to the clipboard and then paste it into a new editing window, all in the time it takes to click your mouse.

Audio Studio retains all of the speed of Sound Forge—there's no waiting for normal cut- or copy-and-paste editing operations. Audio Studio allows you to have multiple editing windows open, copying or cutting and pasting between them quickly and easily. You can also resize each editing window at will. Navigating Audio Studio is easy, with intuitive toolbars for accessing the program's various features. The transport toolbar offers a full array of controls for playback of your file, most of which are duplicated in the editing window.

Zooming is an essential part of precision editing. Like Sound Forge, Audio Studio provides mouse wheel support for zooming, in addition to normal menu choices. Once you've become accustomed to mouse-wheel zooming, you'll probably be intolerant of any program that doesn't support it. Audio Studio comes with an array of menu-selected processes, including Bit-Depth Conversion (choices are 8 or 16 bits), DC Offset Correction, simple and complex Graphic EQ, Fade-In and Fade-Out (in-

cluding Graphic Fade), Insert Silence (with adjustable length), Invert/Flip (polarity inversion), Mute, Normalize, Pan, Resample (to any sample rate between 2000 and 48,000Hz), Smooth/Enhance, Swap Channels, Time Stretch, and Volume (manual adjustment of level).

A large array of effects is also supplied, including amplitude modulation, chorus, delay, distortion, dynamics, envelope, flange wah-wah, noise gate, pitch bend, reverb, and stutter. Looping features are also included, and you can produce music loops in Audio Studio for Sony's ACID looping programs (either the full, professional ACID or the consumer ACID Music Studio). Audio Studio is not intended as an audio restoration program, but it is supplied with Sony ExpressFX Vinyl Restoration module, which I have found to be a very effective tool for removing clicks and pops from LPs (Photo 3). Vinyl Restoration actually removed some nasty clicks on the Ives LP (Table 1) that Groove Mechanic (reviewed in *aX* April 2005) could not take out.

Audio Studio allows you to insert markers and regions into a sound file. Normally, a CD authoring/mastering program such as CD Architect will convert regions into CD tracks and markers into index points. Indexing never became very popular on CDs, and few players support access to them these days. Normally, markers are inserted in the file where each CD track is to occur, plus one at the end.

The professional Sound Forge allows you to automatically convert markers to regions. Marker and region information is embedded into Wave files, which CD authoring/mas-

tering programs can read. Audio Studio does not do the automatic conversion, but you can produce each region manually by marking the beginning and the end of each region (Mark In and Mark Out), and then producing a region. The Regions List window makes the Mark In and Mark Out process fairly easy.

Audio Studio has full recording capability via your sound card's inputs. One essential function of Sound Forge retained in Audio Studio is Extract Audio from CD, which allows you to extract an entire CD and select tracks or a range of data you specify by start and end time. If you extract an entire CD, or specify a range by time, the entire extraction appears in one editing window. The program will insert a marker for each index (or track), a region for each track, or both.

I master all of our concert recordings on DAT. The sound cards on my

**PHOTO 2: The Sound Forge Audio Studio editing environment. The program looks and operates just like the professional Sound Forge, with mouse-wheel support for zooming and easy, intuitive operation.**

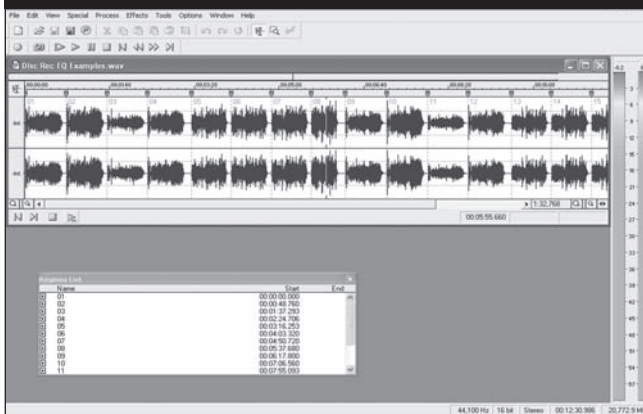
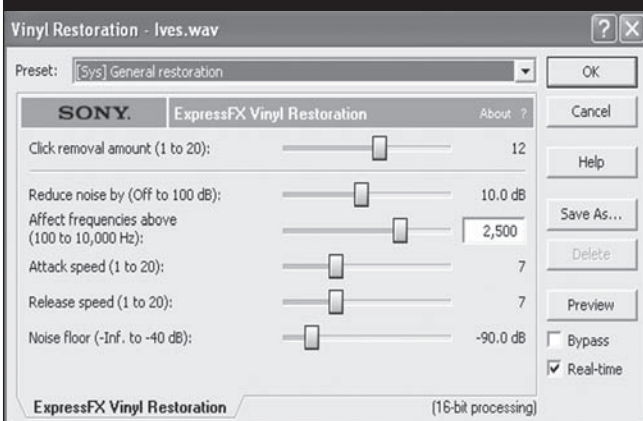
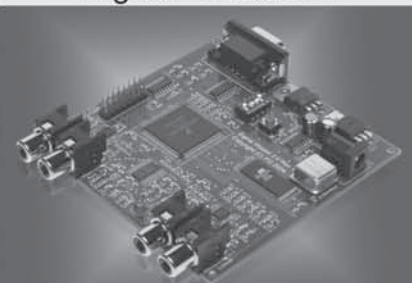


**PHOTO 3: Sony ExpressFX Vinyl Restoration program is one of the best on the market and is supplied with Sound Forge Audio Studio. Six adjustments are included, and the default settings normally yield excellent results.**



## DSP Knowledge...Optional!

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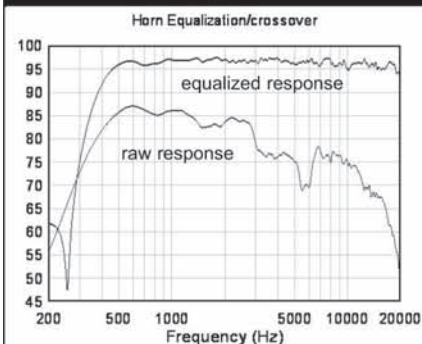
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computers do not have digital inputs, so I normally make a digital copy of my DAT master on a CD recorder. I can then extract the audio from this CD to load it into my editor. This way, the transfer process stays in the digital domain.

There are a number of programs—some of them free—that will extract the contents of a CD, but the ones I've seen store each track as a separate Wave file. This is useless if you need to extract a long piece that has multiple tracks within continuous music. Sony has done it right.

often involves inserting CD tracks in the middle of continuous music, which can be problematic. Track-at-once burners, the most common type, usually insert a 2-second pause between CD tracks, though some allow you to override this.

A seamless joining of multiple Wave files may or may not work, depending on your software and CD burner. Some burning programs support disc-at-once burning, but normally from separate files. Again, I have found achieving seamless joining of files to be a hit-or-miss proposition.

Studio. Both support only track-at-once CD burning. I urge Sony to reconsider this and include disc-at-once burning in these programs.

I also wish that Sony would include a system volume control that would stay “on top” of the program. As it stands, you must use the Windows Volume Control, which hides in the background when you are working in the editor. If you wish to change playback or recording level, you must click on the taskbar to bring up the volume control, which disappears as soon as you go back to editing.

There are a number of other Sound Forge features not included with Audio Studio. Audio Studio does not support DirectX plug-ins, so it is not compatible with Sony Noise Reduction 2.0 package or any other DirectX programs. It doesn't have a playlist/cue list, time display window, effects chainer, clipped peak detection and marking, automated time-based recording, and audio threshold record triggering. Audio Studio has only peak-reading meters; Sound Forge allows you to also choose VU and PPM types.

Sound Forge allows normalizing by the peak or RMS method; Audio Studio uses the peak method only. There's no undo/re-do history in Audio Studio, even prior to closing a file. Sound Forge supports digital audio formats up to 32-bit and 192kHz; Audio Studio supports 8 or 16-bit digital audio at sampling rates as high as 48kHz. There are other differences between them, shown in a comparison chart on the Sony Media Software website at:

<http://mediasoftware.sonymovies.com/Products/showproduct.asp?PID=945&FeatureID=7874>.

Any “light” version of a professional software package is going to eliminate some features found in the full-blown version. Sony, which has generally made sensible choices in designing Audio Studio, is to be applauded for retaining the speed and user-friendly operation of the professional Sound Forge. The boxed package includes a CD-ROM and a 66-page, spiral-bound Quick-Start Manual. You can download a detailed manual that covers both Audio Studio and professional Sound Forge in PDF format from the Sony Media Software website.

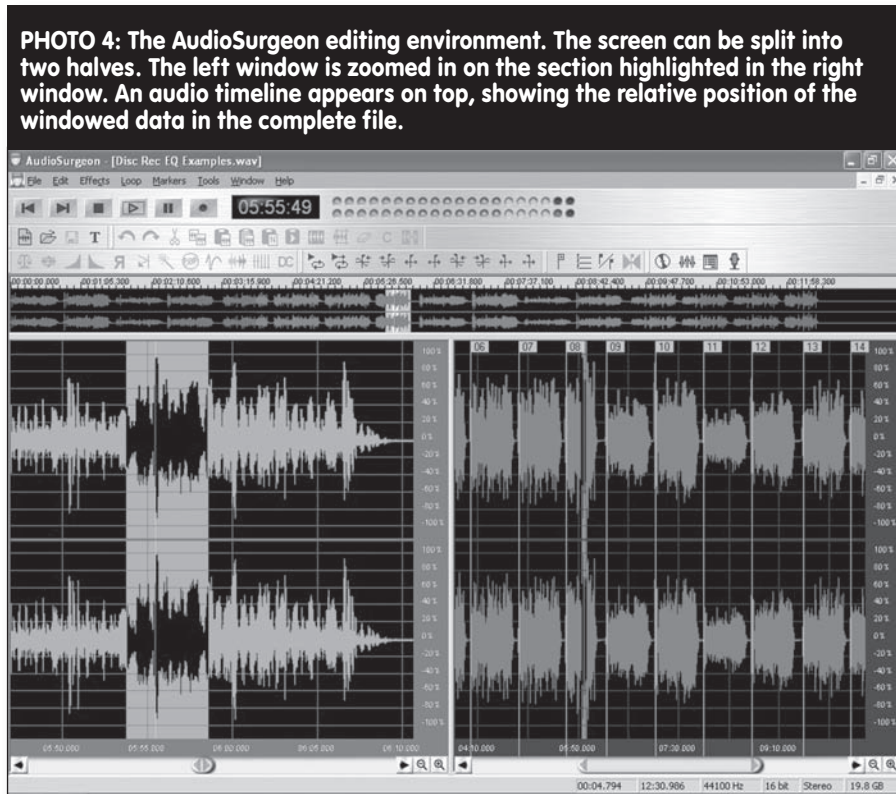


**PHOTO 4: The AudioSurgeon editing environment. The screen can be split into two halves. The left window is zoomed in on the section highlighted in the right window. An audio timeline appears on top, showing the relative position of the windowed data in the complete file.**

Unlike Sound Forge, which lets you save your editing projects in a proprietary .FRG format, Audio Studio does not support non-destructive editing. You can undo and re-do editing operations as long as your file is open. Once saved, your work is cast in stone. Audio Studio allows you to save your work in 22 different file formats, including .WAV and .MP3.

Most CD-burning programs supplied with computers will not recognize markers or regions inserted into Wave files. You need to save each CD track as a separate Wave file, load each file into your burning program, and then burn the disc using the “track-at-once” method. Classical music—especially opera—

Another problem with joining separate Wave files is a doubling of the amount of data on your hard drive. You may choose to keep your original Wave file, with markers and regions in place, plus the separate Wave files you've extracted for each track. This senseless redundancy will bloat your hard drive.

One limitation with Audio Studio is that it doesn't allow you to automatically extract the regions in your file to separate Wave files (Sound Forge does). You'll need to do it manually, by copying and pasting each region to a new editing window, and saving the individual file, a definite inconvenience. I have two other criticisms that apply to both Sound Forge and Audio

## AUDIOSURGEON

AudioSurgeon is a really inexpensive digital audio editor, which you can buy as a download or as a boxed package from many Internet dealers (*Photo 1*). The interface has a very slick, professional-looking appearance (*Photo 4*). Editing is done with normal cut- or copy-and-paste operations, and there's also a Mix Paste function for blending two audio clips. Zooming is done by clicking the + or - buttons in the lower right corner of the screen, but there's no support for mouse-wheel zooming.

AudioSurgeon doesn't boast the lightning speed of Audio Studio. On my home computer, a 2.8GHz Dell Dimension 4600 with 1GB of RAM, loading a 12 1/2-minute file takes about 20 seconds. Deleting 45 seconds of music from that file takes 7 seconds, and restoring it with Undo takes about 2 seconds. On my office computer, which is a 2.4GHz Gateway E-Series, with 384MB of RAM, loading, cutting, and undoing took 20, 20, and 10 seconds, respectively. These processes happen virtually instantaneously with Sound Forge or Audio Studio, on either computer.

Navigating the AudioSurgeon editor is quite easy, and I'm surprised that such an inexpensive program works so well. Only one editing window is available, but you can have several files open at once, cutting or copying and pasting between them. Once each file is opened, you simply go to the Window menu to select the one you wish to view.

Switching between them is fast, once each has been opened. You can split the screen horizontally to show two different portions of the same file, which is also quite useful. The program also has an overview display above the waveform displays that always shows the entire file and highlights the section that is visible in the main waveform display.

AudioSurgeon has an Insert Silence function, which is rather strange, and should really be called "Replace with Silence." With Sound Forge and Audio Studio, you simply put the cursor where you want the silence placed, select the length of the silence in the dialog box, and click the mouse. With AudioSurgeon, the Insert Silence is grayed out until you highlight a portion of the waveform. Then it replaces the portion

you've selected with silence. Everyone doing digital editing will need a conventional Insert Silence function.

Here's a trick: Record one minute of silence with the AudioSurgeon Record Wizard (more on that later). Simply let the recorder run with the mixer levels all the way down. Save this file as Silence.wav. If you wish to insert 5 seconds of silence into your file, open Silence.wav, highlight 5 seconds of it, and copy it to the clipboard. Change the window to the file you're editing, put the cursor where you want the silence, and select Paste. You now have 5 seconds of silence added to your file, right where you want it, rather than replacing a section you wanted to keep.

AudioSurgeon features a basic menu of effects, including scale (manual level adjustment), normalize, fade-in, fade-out, reverse (L & R channels), and echo (reverb). AudioSurgeon also has basic audio restoration tools, including click and pop removal, hiss removal, low pass filtering, and DC offset adjustment. The program has default settings and an automatic mode for each, but you can override these and adjust the parameters manually. Toolbars allow easy access to these functions, in addition to pull-down menu selection.

I didn't find the click and pop removal as effective as Groove Mechanic or Sony's Vinyl Restoration. In general, I also preferred Groove Mechanic's hiss reduction to AudioSurgeon's, but the latter has a sliding noise floor threshold control that is occasionally more useful than Groove Mechanic's three fixed choices. All of these effects run rather slowly.

AudioSurgeon has a Record Wizard that makes recording via your sound card extremely easy. Transport playback controls are available on the toolbar. You can save files in three formats, .WAV, .MP3, and .WMA (Windows Media File). AudioSurgeon supports resolutions up to 16-bit, with sample rates as high as 48kHz. Markers that are stored with the Wave file are easy to insert—for later conversion to CD tracks—and are easy to move around or delete. The program won't extract the contents of a CD.

## DISC-AT-ONCE BURNING

AudioSurgeon has one feature that

makes it unique among affordable editors: it is the only low-cost editor I've seen that supports disc-at-once CD burning. It recognizes the markers you have inserted (and saved) into its own stored Wave files or Sound Forge/Audio Studio Wave files, and turns them into CD tracks when you burn your disc. Some disc-at-once CD burning programs still require you to save each track as a separate Wave file. You must then produce a "play list" or "cue list" of the tracks you want to burn. If everything is working right, the individual files will be joined together properly when the CD is burned.

AudioSurgeon burns a disc from the single, marked file in the editing window, without having to produce a "cue list," a very professional way of operating. Once you've tried real disc-at-once burning, you'll find the old track-at-once method—or any method that requires you to save each track as a separate file—decidedly awkward. Even if you don't use any of the other functions, disc-at-once burning from within the editor is easily worth the \$30 price. You can use Audio Studio as your

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main editor and AudioSurgeon as your CD mastering software. The combined street price of less than \$85 makes this combination a steal.

There is one caveat: AudioSurgeon is unusually finicky regarding the recordable CD drives it will support. I ran into a problem when the drive in my office computer failed. AudioSurgeon had worked fine with the old drive, but not with the replacement supplied by Gateway, a GCC 4481B. Voyetra has a list of supported CD-R and DVD-R drives at: <http://www.voyetra.com/site/products/audiosurgeon/supported.asp>. Unfortunately, this list is very out of date, and the specific model numbers are normally not current.

For example, the list includes the Teac CD-W552E and CD-W552EL, but neither model is made anymore. AudioSurgeon won't recognize the replacement CD-W552G as an external USB2 drive on my office computer. Both the Teac and GCC drives work fine with Sony CD Architect 5.0, the trial version of Adobe Audition 1.5 (which also does disc-at-once burning without splitting up your file), Roxio E-Z CD Creator, and the OEM version of Nero Express 6 (Nero came free with the Teac drive).

I raised this issue with Voyetra's tech support folks, and was simply told that if the drive is not on this list "it won't burn." They won't answer any questions about newer drive models not on this list, nor about what is specifically required of a drive in order to work with the program.

In answer to one of my inquiries, Voyetra tech support noted that since all computers come with burning software, the fact that AudioSurgeon's burner did work should not be an issue. They really seem clueless regarding the importance of their disc-at-once burning for some customers. This is very frustrating, so I strongly suggest downloading the trial version to see whether it is compatible with your drive. If your current drive fails, you will likely be out of luck.

If, for some reason, you choose to store each CD track as a separate file, AudioSurgeon supports this. Simply select Split Markers Into Files to display a dialog box that allows you to select as many or as few tracks as you like. If you're unsure which tracks you

want, you can select a track for listening within this dialog box. This is another extremely useful feature, rare in such an inexpensive product.

Another nice feature of AudioSurgeon is the Voyetra Mixer, which you can use instead of the Windows Volume Control. It has many customizable options, including an "Always On Top" choice that puts it on top of your editing application. It also works fine with Sound Forge and Audio Studio.

If you download asurgeon.exe, you will get AudioSurgeon version 6.20, but this version will not accept product ID codes supplied with the boxed product. Most Internet computer dealers selling the boxed product are still shipping version 5, which is out of date. In order to update this version to version 6 for free, go to Voyetra's FTP site <ftp://ftp.voyetra.com/pub/voy/asurgeon>. Download the file dr\_50enc.exe, which, contrary to the filename, is really version 6.01.2172.

The Voyetra website notes: "If you already have the previous version 6.01.2172 of December 23, 2003, do NOT update to the newer version, as there are NO new features except for Copyright/Licensing changes. You need to make sure that you are using the correct Product ID to unlock the appropriate version."

Voyetra has added further confusion to the version issue with its installer, which tells you that version 5 is being installed, even though it is really installing version 6. Once installed, you can go to "Help" and "About AudioSurgeon" to verify installation of the correct version. I see no need to order the boxed version, which is actually a couple of dollars more than the download, and won't allow you the benefit of a free trial. The 20-page manual supplied with the boxed version is the same one you can download as a .PDF file, but the download version has color screenshots (the printed manual is black and white).

## DC SIX

DC Six is an audio restoration, audio editing, and audio enhancement software package selling for under \$200 (Photo 5). DC stands for Diamond Cut Productions, Inc., the designer of this package and its predecessors, DC Five

and DC 32 ([www.enhancedaudio.com](http://www.enhancedaudio.com)). Enhanced Audio, distributor of DC Six, is a partnership between Diamond Cut Productions and Tracer Technologies ([www.tracertek.com](http://www.tracertek.com)). Tracer sells a variety of audio gear including sound cards, phono preamps, record cleaning products, microphones, and mixers.

Diamond Cut audio restoration software was written by Craig Meyer and Rick Carlson, two engineers working on the restoration of old sound recordings at the Edison National Historical Site in West Orange, N.J. A companion product, DC Live/Forensics, is intended for forensic audio uses, and allows live "on the fly" audio processing. One manual covers both programs, but I did not evaluate the Live/Forensics package. DC Six version 6.14 is the most recent upgrade, containing a number of bug fixes for users of earlier versions. Registered users can download the 6.14 patch at <http://www.tracertek.com/software1.htm>.

Enhanced Audio claims that DC Six is "the world's most powerful audio restoration software." Tracer Technologies calls it "the world's best audio restoration/enhancement and editing product," and further notes "You won't find a better audio editor, audio restoration and audio enhancement package anywhere in the world. Period."

During this evaluation I did considerable comparison with my reference editing and restoration programs, Sony Sound Forge 7.0 and Noise Reduction 2.0 packages. Street price for these two programs is around \$550, nearly three times what DC Six costs (remember that Sound Forge comes bundled with CD Architect 5.0). But, in view of the manufacturer's claims for DC Six, the comparisons seem fair.

Their claims also seem to invite comparisons with five-figure editing and restoration programs such as Sadie, Cedar, and Sonic Solutions. The Sadie Disc Editor combined with Cedar audio restoration software is considered the state-of-the-art in audio editing and restoration for the Windows platform; Sonic Solutions holds that same position in the Macintosh world. Several of my friends in the recording and restoration business use both of these products.

DC Six comes with a full array of

programs for click, crackle, and hiss removal, equalization, and a host of effects (*Photo 6*). It was designed for restoring every conceivable type of recording, including cylinders, lateral and vertical 78s, 45s, broadcast transcriptions, LPs, and tapes.

Noise reduction modules include EZ Clean, EZ Impulse Noise, Expert Impulse Noise, Continuous Noise (Hiss), Harmonic Reject Filter, and a Dynamic Noise Filter. The Impulse Noise filters are intended for click and crackle removal. The EZ versions are for novice use. EZ Impulse noise has only two sliding controls—one for scratch and one for crackle.

The Expert version has separate slider adjustments for threshold, size, and tracking. EZ Clean has three controls, for scratch, crackle, and hiss. All of these programs include many presets (a few dozen on some) for various types of source material, including specific types of 78-rpm and LP records, and a variety of tape sources.

Most affordable hiss removal programs use the spectral subtraction method. The DC Six Continuous Noise

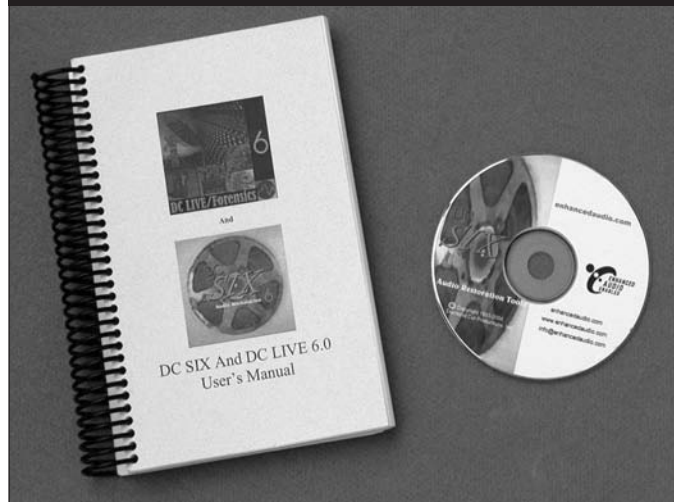
processor also has two other choices—Normal CNF and Audio Spectrum CNF. The Audio Spectrum CNF mode works “on-the-fly” and doesn’t require sampling the noise first. Every module included with DC Six allows previewing, so you can play the file and adjust the controls to hear the effects. Once you’re satisfied, you can process the entire file.

DC Six also has a variety of filters, including Lowpass, Bandpass, and Highpass, all with adjustable slopes and a choice between Butterworth and Chebyshev (their spelling). There’s also a Notch Filter with adjustable bandwidth and “Q,” plus median and averaging filters. Three equalizers

are also included: 10-Band Graphic, 20-Band Graphic, and a terrific Paragraphic equalizer, which allows you to literally draw the filter shape. All of these filters have a multitude of presets and a preview mode.

One of the Paragraphic presets is an RIAA de-emphasis curve. Tracer Tech-

**PHOTO 5: DC Six packs a toolbox of audio editing, restoration, and effects programs at an affordable price. The program comes with a 400-page manual that requires considerable study in order to master its operation.**



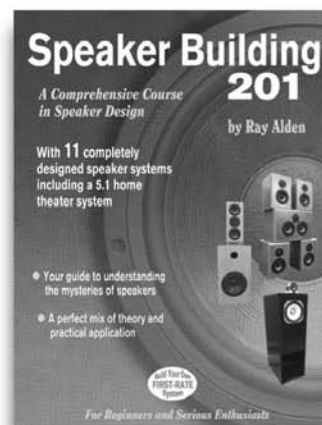
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nologies sells two flat phono preamplifiers (which don't compensate for the RIAA curve). After making a flat transfer of an LP to your computer, you can use the Paragraphic's RIAA setting to add RIAA playback EQ in the digital domain. Enhanced Audio believes that this is a more precise method than conven-

tional analog RIAA phono preamps.

One question I posed to Enhanced Audio concerned the phase characteristics of the digital equalization. Digital filters can be designed with perfect group delay characteristics and can provide frequency equalization without the nasty phase shifts of analog filters. But, RIAA

phono preamps compensate for both the amplitude and phase characteristics of the record, as played by a magnetic (velocity-sensitive) pickup. No one ever considers the phase characteristics of the RIAA curve because with analog filters, the phase correction comes along for the ride when the correct frequency equaliza-

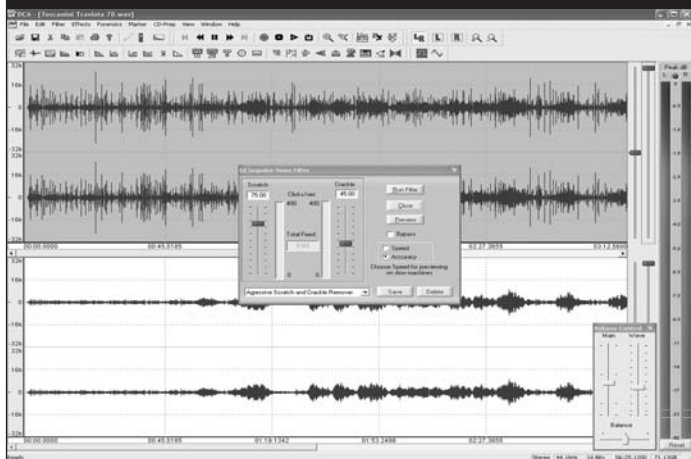
tion is achieved.

In *Fig. 1*—two graphs I generated using computer simulations—the top graph shows the familiar RIAA emphasis and deemphasis curves, i.e., the frequency response. The bottom graph is the phase response of the emphasis and deemphasis networks. In both cases, the complementary curves yield a flat response.

If RIAA de-emphasis is done in the digital domain, it must work exactly like an analog filter in order to provide correct amplitude and phase equalization. I sent several e-mails to Enhanced Audio's technical support asking whether or not the paragraphic RIAA EQ provided proper phase response, but they never answered the question (all of my other questions were answered, however). I would not do RIAA digitally unless I was certain that the phase response was correct. The RIAA filter is "phase neutral" and provides zero phase shift.

DC Six also includes an array of studio effects, including reverb, echo, a dynamics processor, a punch and crunch multi-band dynamics processor, and a channel blender. There's even a very

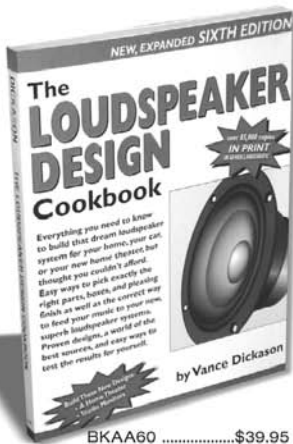
**PHOTO 6: The DC Six environment. The screen shows the program in the "classic" editing mode, with the source window on top and the destination window on the bottom. Processed files appear in the destination window. The EZ impulse noise filter is open in the center of the screen, and the "always on top" volume control is in the lower right.**



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flexible virtual valve amplifier for adding tube warmth or sweetness to a recording. The change speed processor allows fixed speed changes, such as an analog pitch control, over three ranges—1%, 10%, and 100%. This allows you to make any conceivable correction to an old recording that has been transferred at the wrong speed (remember that most so-called 78-rpm recordings were actually recorded at some speed “in the ballpark,” from below 70 to the mid-80s, and sometimes even higher).

You can also set this processor to adjust pitch over time, to correct “pitch creep” in a recording. There are a variety of presets, including corrections needed for fractional speed mastering. Stretch and Swish is a DJ-type feature, allowing you to change tempo while maintaining the same pitch. All of these effects allow previewing.

DC Six has an incredibly versatile file conversion feature, which is an elaborate “Stereo-Mono” switch. You can convert a mono file into stereo, with a variety of “fake” stereo settings, or a stereo file into mono L+R, mono L-R, left only, right only, stereo reverse, and also

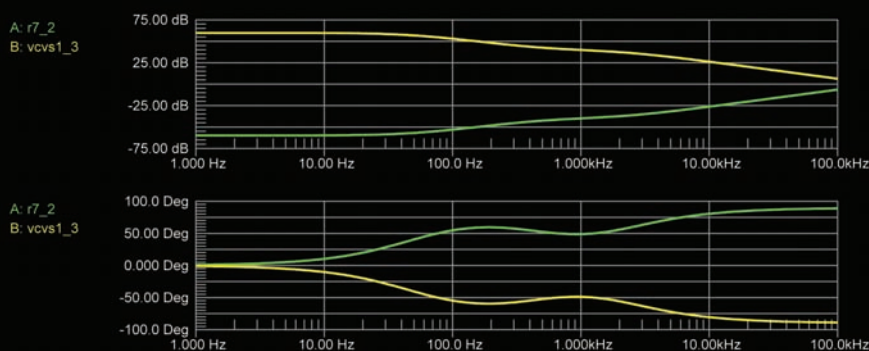
$L = L + R$  and  $R = L - R$ . These are especially useful if you’ve stereo-transferred a mono 78-rpm disc.

You can combine the two groove walls of a laterally-cut or vertically-cut recording optimally. There are separate level adjustments of the two channels and a time delay adjustment. In both cases, the mouse is used for coarse adjustment

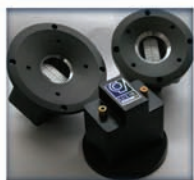
and the arrow keys on your keyboard allow fine-tuning. Again, many presets and a preview mode are included.

Many restoration engineers prefer to do stereo transfers of mono records and tapes. Later, in the digital domain, phase and level corrections can be made for optimum sum of the two channels to mono. You can use the file conver-

**FIGURE 1: The RIAA recording and playback characteristics deviate from a flat response both in amplitude and phase, but phase is seldom considered because it happens automatically with analog filters. The top graph shows the familiar RIAA frequency response, and the bottom graphs show the phase response. The yellow traces are the recording curves (the record played by a magnetic transducer), and the green traces are playback. The graphs were generated by computer simulations.**



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DEQX PDC 2.



sion tool for this in conjunction with the X-Y vector display, also known as a Lissajous pattern (Photo 7). The preview mode allows real-time adjustments while viewing the results on the X-Y display, before processing the entire file. You can also use the X-Y vector display to check whether a monaural recording has been issued in fake stereo (you can play a CD track from within DC Six's CD ripping window).

If you've done a stereo transfer of a mono source, split and recombine allow independent processing of the left and right signals, before putting them back together. Both the file conversion and the split and recombine are incredibly powerful tools normally found on much more expensive restoration software. On a future release, they might consider a file conversion setting that converts  $L = L + R$  and  $R = L - R$  stereo into normal L and R stereo. This would be useful for decoding an MS recording in the digital domain.

Two forensic tools are also included. A De-Clipper can reduce distortion on clipped waveforms. Presets are included for various types of clipping, including digital waveforms, analog tape, and various types of amplifiers. Normally, you must first reduce the overall level of the file before the program can eliminate the flat-topping. You can configure a brick wall filter as high-pass, low-pass, bandpass, or band-stop. DC Six includes a multi-filter function, which is really a chainer that lets you daisy-chain multiple processors together.

## DC SIX EDITOR

The DC Six editor is very fast, but not nearly as easy to use as Sound Forge or Audio Studio. You can't zoom with the mouse wheel; you need to use the zoom buttons on the toolbar, or select them from the pull-down menu. If the file has been zoomed so that only a portion of it fits on the screen, the DC Six editing window doesn't scroll during playback. Once the cursor reaches the

end of the visible portion of the file, it goes off the screen and disappears. With Sound Forge and Audio Studio, the editing window scrolls the waveform so the playback cursor is always on the screen, which is much more sensible.

DC Six will store its own markers in the Wave file, but it does not recognize markers and regions embedded in Wave files by Audio Studio, Sound Forge, or AudioSurgeon, and those programs won't recognize DC Six's markers. It's unfortunate that DC Six doesn't use the professional, industry-standard file marking system supported by these

The advantage of the fast mode is increased speed, but there's a danger of overwriting your original file with the processed file unless you do a "Save As" after processing. The source file and destination file concept of classic mode ensures that your original file is not changed. But, if you do any editing on the source file and then close the file, it will be automatically saved without prompting you. I don't understand the logic behind this.

The full-screen display in either mode may be uncomfortably large on some screens. You can re-size it in a window, but the re-sizing is not retained with the file. It will default to full screen every time you open it.

You can use an undo command when you don't like an edit or a signal process you performed. There's no redo command, but there are ways to redo. For any filter or process operation, you can run the filter or processor with the same settings as before. If you're in the fast edit mode, there's a history window that will allow you to redo anything that you undo.

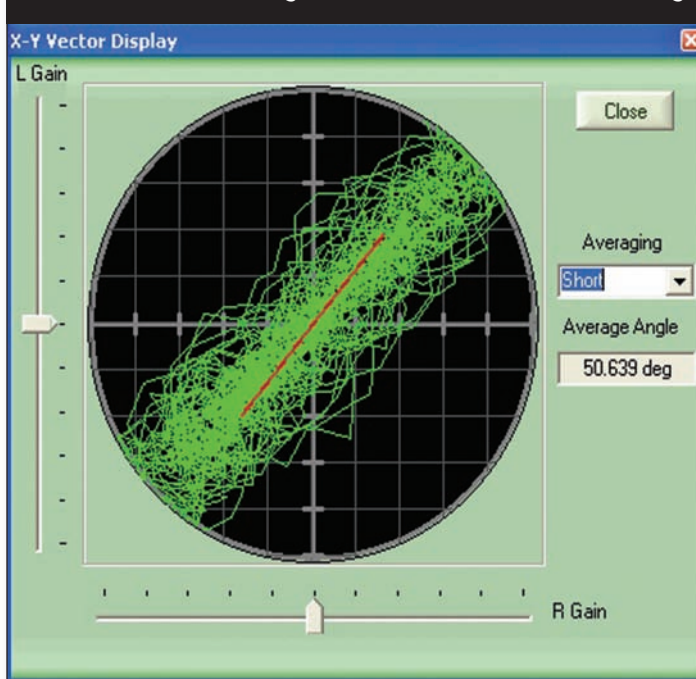
In the classic mode, there's no way to redo edits, because the history window doesn't work in the classic mode. This is also strange, because the classic mode is supposed to be the preferred method for editing

and processing. Both Sound Forge and Audio Studio have redo commands on the edit menu.

DC Six includes its own volume control for playback monitoring. Once you open it from the view menu, it stays on top of the program at all times. I find this a really nice feature, one I wish Sony would incorporate into the Sound Forge and Audio Studio programs.

DC Six supports DirectX plug-ins, even though its own processing programs are not DirectX-compatible. I tried using my Sony Noise Reduction 2.0 DirectX programs within DC Six and couldn't get them to work. This is not really an issue, because I'd much rather run them in the Sound Forge editor. The

**PHOTO 7: DC Six's X-Y Vector display is an extremely useful tool that you can use for azimuth adjustments, stereo-to-mono-file conversions, and checking for fake stereo on a mono recording.**



other programs. DC Six has a paste silence function, which is basically the same as insert silence. You can paste a silence of any length to the beginning or end of a file, or the beginning of a selected area. This last is a bit different from Sound Forge and Audio Studio, but it works.

There are two edit modes—classic and fast. In the default classic mode, the original file opens in the top window. After that file has been processed, the processed version appears in the lower window. In fast mode, there's only one window, which fills the whole screen vertically and changes from the original file to the processed version after processing has been done.

DC Six restoration programs are not DirectX compatible. DC Six will extract the contents of a CD, but each track is saved as a separate Wave file. DC Six also supports recording via your sound card, with resolutions up to 32-bit and sampling rates up to 192kHz.

DC Six doesn't burn CDs. After you add markers to a file, you select Chop File Into Pieces from the CD Prep menu, which breaks your file up into separate Wave files. Then, you use CD burning software of your choice to burn your CD. Because DC Six's markers aren't recognized by other programs that do disc-at-once burning, it is not suitable for serious classical music editing.

## RESTORATION RESULTS

Table 1 is a list of recordings I transferred for evaluation of audio restoration programs. Since my computer is in a different location from my main audio system, I transferred all recordings to a CD-R using a Sony CDR-W66 professional CD recorder and extracted the tracks to individual files using Sound Forge. I have considerable experience with restoration of these recordings with Sound Forge 7.0 and Sony's companion Noise Reduction 2.0 package.

DC Six did a generally decent job on most vinyl LP scratches. One disc that gave it some trouble was the Ives. This LP has a series of once-around clicks due to a pressing defect, lasting about two minutes. The vinyl restoration module included with Sound Forge and Audio Studio is the only program I've used that removes them all in one operation. A barely audible noise is left in place of the largest clicks.

With DC Six, there's no single pass approach that will fix this file. You must use manual interpolation to remove each click. Zoom in on each one, highlight it, and hit "i" on your keyboard. Then you can run E-Z Clean to remove the remaining crackle.

I sent the most difficult portion of this file to Enhanced Audio's tech support to see what they could do with it. Their cleanup was not as effective as what I achieved with one pass of Sony Noise Reduction 2.0 using the default settings and no manual operations. The DC Six restoration left noises in place of the large clicks and was not as effective in removing the remaining low-

level crackle as Sony Noise Reduction 2.0.

In general, I found the E-Z Clean to be the best method for vinyl click removal in DC Six. I also needed to use the interpolation on the large click in the Furtwängler Beethoven 9th excerpt. Again, Sony Noise Reduction 2.0 took it out with one pass on the default settings. Neither program was completely successful with the *Belshazzar's Feast* LP I deliberately damaged. The Hindemith and Wagner discs, with light-to-moderate random ticks, fared well on both programs.

On 78-rpm material DC Six is capable of good results, but you must be prepared to do a lot of fiddling. On the Toscanini Traviata Prelude, Sony Noise Reduction 2.0 click and crackle remover did an excellent job with this very crackly disc using the program's default 78-rpm setting. The various presets in DC Six are just a rough starting point, and much user intervention is required beyond that. With DC Six, you must spend a lot of time and do a great deal of adjusting to achieve satisfactory results, but I was never able to get the *Traviata* Prelude as quiet as the Sony-processed file.

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The advertisement features a stylized illustration of several vacuum tubes of various sizes, some with glowing filaments, set against a background of red and yellow stars. The text is in a bold, retro font.

I also found that DC Six added a very subtle texturing to the high string sound in the *Traviata* Prelude. The sound with Sony Noise Reduction 2.0 is silky and smooth after processing—sim-

ply more natural. Similarly, on the Caruso “Over There” recording both the primitive orchestral accompaniment and the voice sounded more like the unprocessed record with Sony Noise Re-

duction 2.0. It was difficult to achieve a satisfactory reduction in crackle at the beginning of the Toscanini *Midsummer Night's Dream* Scherzo with either program without compromising the music.

I have two friends in the audio restoration business who do tape hiss reduction using Cedar and Sonic Solutions equipment and software. Both have said that 6 to 8dB of hiss reduction is the most that can be achieved without audible side effects. What's interesting is that I found exactly the same thing to be true of Sony Noise Reduction 2.0 program.

The first side effect, if you go too far, is an audible reduction of high frequencies. You can often compensate for this with digital equalization included with Sound Forge. If you go too far, you will get audible artifacts such as high-frequency swishing sounds.

I used DC Six's continuous noise filter (CNF) to reduce tape hiss on the excerpt from the Fauré *Requiem* conducted by Nadia Boulanger. You normally begin hiss reduction by taking a sample of the noise where the music is silent. Although the manual doesn't make this clear, the spectral subtraction mode was designed primarily for speech or very noisy recordings.

On this recording, using the spectral subtraction mode with even 6dB of hiss reduction produced audible artifacts and strange swirling sounds. The Normal CNF mode worked much better. The 15ips reel-to-reel default settings include 35dB of noise reduction, which is way too high for most applications.

With more realistic levels of noise reduction—10 to 12dB—DC Six did a very good job with this file. I wonder whether the calibration for the amount of attenuation is accurate in DC Six. Subjectively, Sony Noise Reduction 2.0 seemed to achieve lower levels of hiss for a given decibel amount of noise reduction than DC Six, so I'm not sure the calibrations for the two programs are comparable. DC Six certainly did a good job with hiss reduction on this example. Groove Mechanic uses the spectral subtraction method, but it also did a fine job with this excerpt using the most conservative setting, as I noted in my review of that program.

The Boulanger recording is quite

## TABLE 1: RECORDINGS USED TO EVALUATE DC SIX 6.0

### LP Clicks and Pops

Charles Ives: *The Things Our Fathers Loved* - “Slow March,” “Canon,” and “There's a Certain Garden.” Helen Boatwright, Soprano; John Kirkpatrick, piano. Columbia Masterworks M4 32504, Record 3, Side 1, Band 1. (Loud crackles and pops at the beginning due to a pressing defect.)

Beethoven: Symphony No. 9 in D minor. Berlin Philharmonic Orchestra conducted by Wilhelm Furtwängler, Rec. London 1 May 1937. Toshiba/EMI WF-60073-74. Excerpt from 2<sup>nd</sup> Mvt. (There's a gouge in these otherwise pristine records on Side 2, causing two ticks, one minor and one very loud.)

Wagner: *Tristan und Isolde* - Prelude to Act 1. Los Angeles Philharmonic Orchestra conducted by Erich Leinsdorf. Sheffield Lab LAB-7 (Direct-to-Disc). (Light, random ticks.)

Hindemith: *Symphonie Mathis der Maler* - Mvt. 1 Introduction. Boston Symphony Orchestra conducted by William Steinberg. Deutsche Grammophon 2530 246. (Moderate, random ticks.) (An outstanding performance by an underrated conductor that deserves a CD re-mastering by DG.)

Walton: *Belshazzar's Feast*. Crane Chorus and Crane Symphony Orchestra conducted by Brock McElheran. CSM-1003, a record produced privately by The Crane School of Music, SUNY Potsdam, from analog tapes I recorded in 1984. (I had some extra copies of this record lying around, so I sacrificed one for de-clicking tests; I put three large, evenly-spaced scratches at the beginning of the piece, and three more large scratches 1/2-inch apart at the beginning of Side 2.)

### Tape Hiss

Fauré: *Requiem* - “Libera me.” Crane Chorus and Crane Symphony Orchestra conducted by Nadia Boulanger. Spring Festival of the Arts, 1958. 15-ips full-track mono original tape.

Respighi: *The Birds* - “Prelude.” *The Pines of Rome* - “Pines of the Villa Borghese.” London Symphony Orchestra (Birds) and Minneapolis Symphony Orchestra (Pines) conducted by Antal Dorati. Mercury CD 432 007-2. Digital re-mastering from original 15-ips, 3-track analog tape. No digital de-hissing was done on the Mercury CD transfers, so there is some residual tape hiss on this recording.

**Clipping Tests** (These LPs were deliberately transferred to digital at excessively high recording levels, resulting in clipped waveforms on peaks.)

Stravinsky: *Firebird Suite* - “Infernal Dance.” Atlanta Symphony Orchestra conducted by Robert Shaw. Telarc DG-10039.

Wagner: *Siegfried* - Act I Conclusion (“Forging Scene”). Wolfgang Windgassen, tenor (Siegfried); Gerhard Stolze, tenor (Mime); Vienna Philharmonic Orchestra conducted by George Solti. Decca (Germany) 6.35252.

Wagner: *Die Walküre* - Opening of Act II. George London, bass-baritone (Wotan); Birgit Nilsson, soprano (Brünnhilde); London Symphony Orchestra conducted by Erich Leinsdorf. Decca (England) 7BB 125-9.

**78-rpm Tests** (These discs were transferred to digital using a Technics SP-15 turntable, SME-3012R tonearm, Stanton 500AL cartridge with truncated elliptical styli of varying sizes, and my modified McIntosh C-8 preamplifier. Playback speeds and phono equalization are noted for each record. A modified Phoenix Systems Parametric Equalizer was also used for the acoustical recording.)

Mendelssohn: *A Midsummer Night's Dream* - Scherzo. New York Philharmonic Orchestra conducted by Arturo Toscanini. Brunswick 50074-A, rec. 1926. 78.26-rpm, 400Hz bass turnover, no treble rolloff. (Clean copies of this record are hard to find—mine has severe crackle at the beginning and a number of ticks and pops.)

Verdi: *La Traviata* - Prelude to Act I. Philharmonic-Symphony Orchestra of New York conducted by Arturo Toscanini. Victor 6994-A, rec. 1929. 77.0-rpm, 300Hz bass turnover, -5dB at 10kHz. (An unworn copy of a late-1930s pressing, but there are a large number of clicks and pops, plus the usual steady-state surface noise.)

Cohan: “Over There.” Enrico Caruso, tenor; Victor Orchestra conducted by Josef Pasternack. Victor 87294, rec. 1918 (acoustical recording). 75.00-rpm, flat phono EQ, parametric equalizer set to +6dB at 180Hz, 0.7 octaves. (A pristine, single-faced copy with virtually no clicks or pops, but the usual steady-state surface noise from the abrasive fillers used in Victor's shellac formula.)

Verdi: *Il Trovatore* - “Miserere.” Rosa Ponselle, soprano (Leonore); Giovanni Martinelli, tenor (Manrico); Victor Orchestra conducted by Giulio Setti. Victor 8097-A, 77.0-rpm. 400Hz bass turnover, flat treble. (The usual steady state surface noise present on scroll-label Victors from the late 1920s, plus some clicks and pops, and distortion on the final chord caused by too high a recording level.)

primitive, with very little real treble, which is why I was able to achieve more noise reduction without side effects than I would on high fidelity material. For a real test, I tried hiss reduction on two tracks from Mercury's *Dorati Conducts Respighi* CD. Mercury normally put recording tape, rather than leader tape, complete with room tone between movements, so it's easy to find places to get a sample of the hiss. I decided to push both DC Six and Sony Noise Reduction 2.0 a bit by doing 12dB of hiss reduction on the two tracks from this disc.

I generally preferred the processing with Sony Noise Reduction 2.0, which softened the treble a bit and also truncated the hall ambience slightly. With DC Six, there was greater reduction of hall ambience, and the soundstage shrunk both left-to-right and front-to-back. DC Six also added a bit of artificial texture to the strings, particularly the violins. The Sony-processed file sounded closer to the original.

DC Six's De-Clipper works quite well on files with short-duration, moderate clipping. With digital clipping, you must reduce the overall level of the file by 3dB in order to give the program some headroom to rebuild the peaks. The transients on the "Infernal Dance" from the Telarc *Firebird* were rebuilt quite effectively, not exactly like the originals, but reducing the offensive part of the distortion.

The clipped peak restoration module in Sony Noise Reduction 2.0 produced slightly cleaner results, however. Like Sony Noise Reduction 2.0, DC Six won't salvage a file with severe, continuous clipping. The program must have leading edges from which to rebuild peaks.

I generally preferred the various noise-reduction programs supplied with Sony Noise Reduction 2.0, both in terms of performance and ease of operation. The DC Six package offers good performance for the money, though. Beyond the two "E-Z" noise removal programs, the learning curve is rather steep.

Like any program of this type, results will depend a lot on operator skill. The manual is over 400 pages long. About a quarter of it is a glossary, but you'll probably need to read most of the remainder, and do a lot of experiment-

ing, to become really proficient with the program.

## CONCLUSION

Sound Forge Audio Studio is—hands down—the best low-cost editor I have seen, and at the street price of around \$52 it's a steal (try [www.ecost.com](http://www.ecost.com)). It combines the speed and ease of use of far more expensive programs with a well-chosen set of features, including an effective vinyl restoration module. Its biggest limitation, like professional Sound Forge, is the lack of disc-at-once burning. I hope Sony will add this in the future.

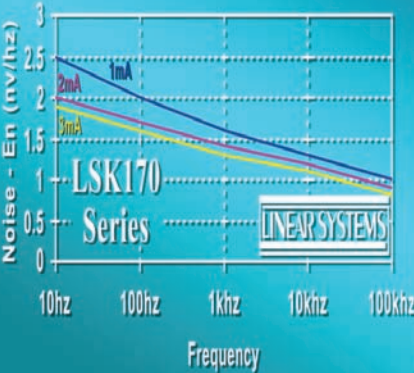
At around \$30, AudioSurgeon provides a decent set of basic editing functions, but editing and processing functions are rather slow. Its built-in disc-at-once burning, which recognizes industry-standard Wave file markers, makes it an invaluable program if it will work with your CD burner. It's easily worth the money even if you use it only as your CD burning software. Combine it with Sound Forge Audio Studio, and for less than \$85 you'll have a pretty serious editing/burning system at an

amazingly low price. Improved support for recordable CD drives is sorely needed, and can only help Voyetra's sales.

DC Six combines a basic editor with a large number of audio restoration tools and effects for a very reasonable price, more than you would expect for \$199. The editor is rather cumbersome compared to Sound Forge or Audio Studio, and Sony Noise Reduction 2.0 package offers superior results and easier operation for vinyl restoration and 78-rpm noise removal. The DC Six package is a lot less money than the combined cost of Sound Forge and Noise Reduction 2.0, and many of the modules included with DC Six are not available with the Sony programs. Some, such as DC Six's file conversion and X-Y Vector display, are extremely powerful, and normally found only on much more expensive software. Your choices will ultimately be determined by your specific needs and budget.

By the time you read this, Sony will have introduced Sound Forge 8.0, which will be bundled with CD Architect 5.2, for a preliminary combined price of \$299.96. If you are a teacher or a stu-

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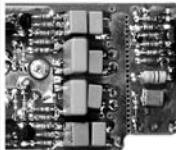
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#### Manufacturer's response:

I was impressed with the thorough nature and think that overall, this review was a pretty positive experience. With that being said, there were a few issues both technical as well as some general comments on the review.

If this was an ad for Sound Forge, I think it would be wonderful, but if you're comparing three products, probably a power user of one is not the best person to review the other two. I don't believe one point was made either positive or negative about our product without the mention of how much better Gary felt about Sound Forge. I'm very happy for him. He likes Sound Forge ... been using it for years ... that's great.

Here are a few things that users may not know. First, the average age of an Audio Restoration and Enhancement customer is 50+. Not all of these individuals are computer savvy. When they have a problem ... who do they call? With EnhancedAudio, they call us, we answer the phone and talk them through it. Whether that call takes an hour or 10 minutes, our customers know that they can count on us to support the product. When a customer of Sound Forge, who isn't a veteran of the product calls Sony, who do they talk to? My guess ... a voicemail telling them to get out the old credit card. And they paid about \$200+ more for that privilege. This should be mentioned, as support is a real issue until you become the power user that Gary is ... of any product.

With regard to automatic saving, in classic mode, everything is saved without prompting. This is one of the reasons that average folks without several years of Sound Forge experience can use it safely. Everything is also reversible to any degree.

Also, you are right that there is no redo for "edits" in classic mode—cuts, pastes, copies, for example. Our theory being that if you did an incorrect cut, you'd undo it and why would you want to do it again incorrectly? And, for lots of editing functions, we do recommend Fast Edit mode as the correct choice (the

article states that we recommend classic for both processing and editing).

Again in classic mode—any processing function (one that produces a destination file) can be redone either with the same settings or with new ones. Once processing is done, it is not necessary to close your filter window to listen to the resulting file. If you don't like it, just click run filter again and it'll be done again.

Next, it's true we will normally end up with separate wave files when done. We recommend this as the way for folks to use the program. However, this does in no way mean you can't do disk at once. Just load our wave files into Nero and tell it to burn DAO. You can also, depending on the burning program, specify the amount of silence between tracks. In longer classical pieces that you want to split into files, just choose Zero seconds. There's no problem that we are aware of in producing CDs mastered any way you want using a good program like Nero.

Finally, spectral subtraction filters will produce artifacts at much more than 6dB. That's why it's not the default setting for DC Six. We call the Normal CNF normal because it's a much better approach allowing greater degrees of continuous noise reduction without as much artifacting.

By the way, this filter is flexible enough in its internal design to allow it to remove the last traces of crackle as well. The amount of noise reduction is displayed accurately we believe.

Jeff Klinedinst

*Enhancedaudio.com Inc.*

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