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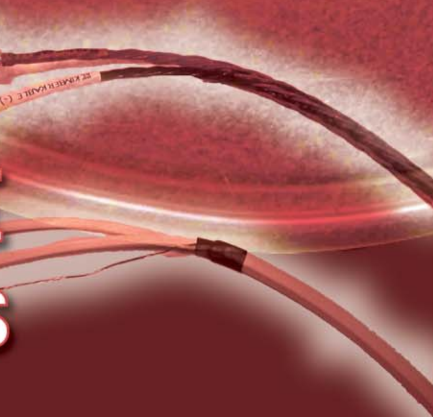

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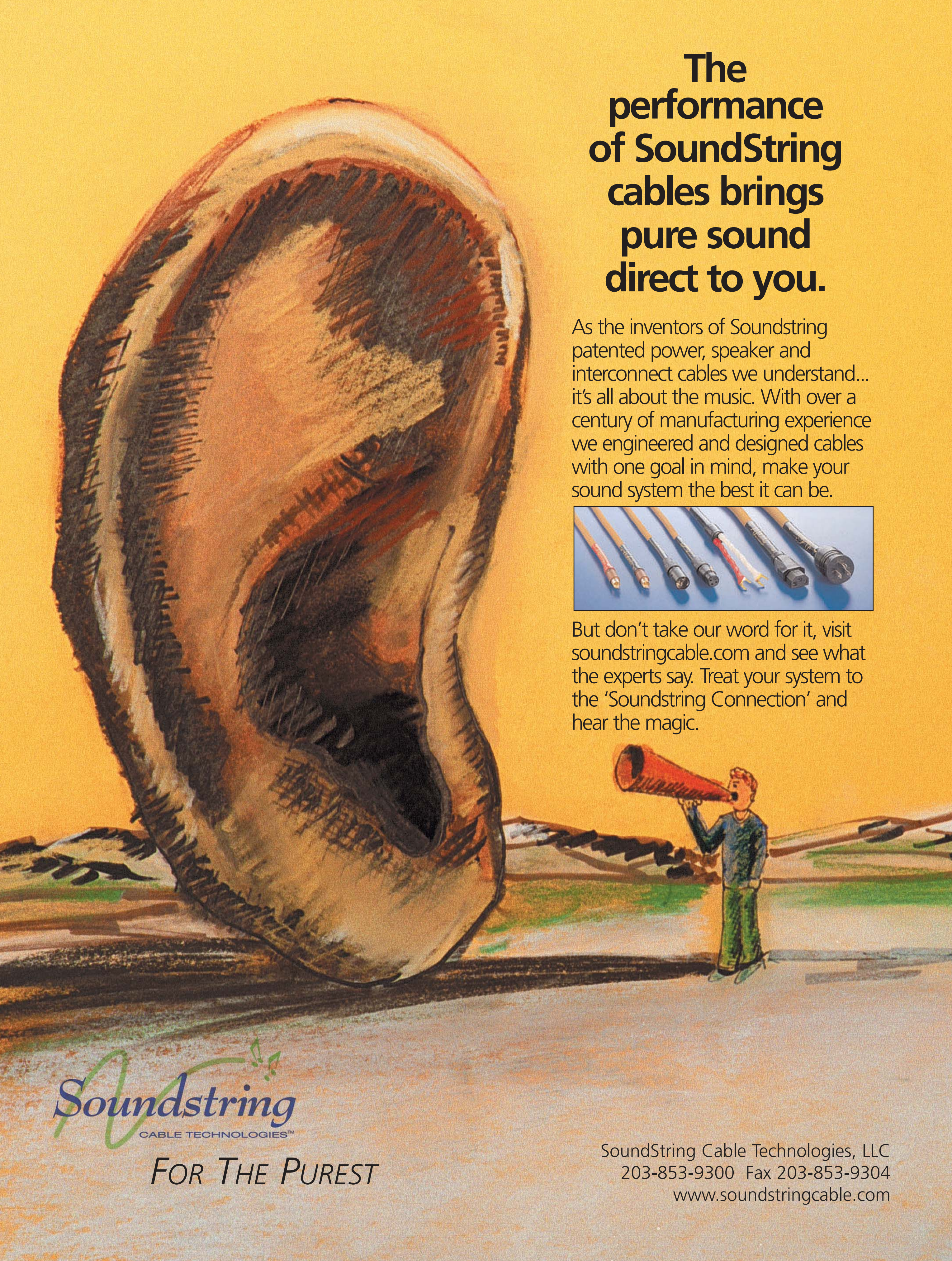
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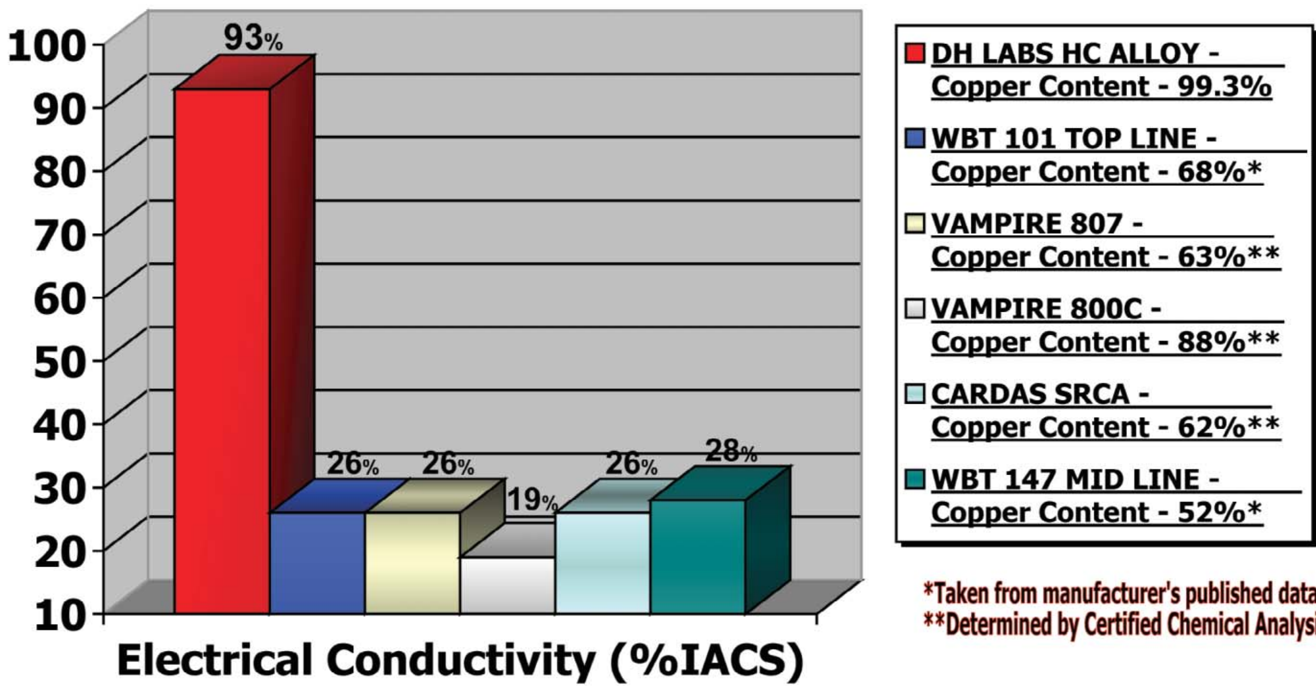

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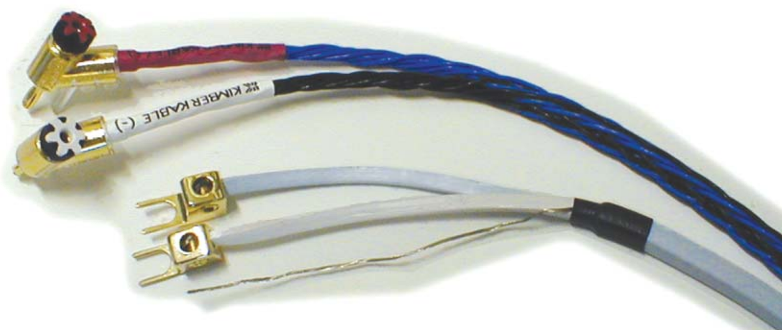

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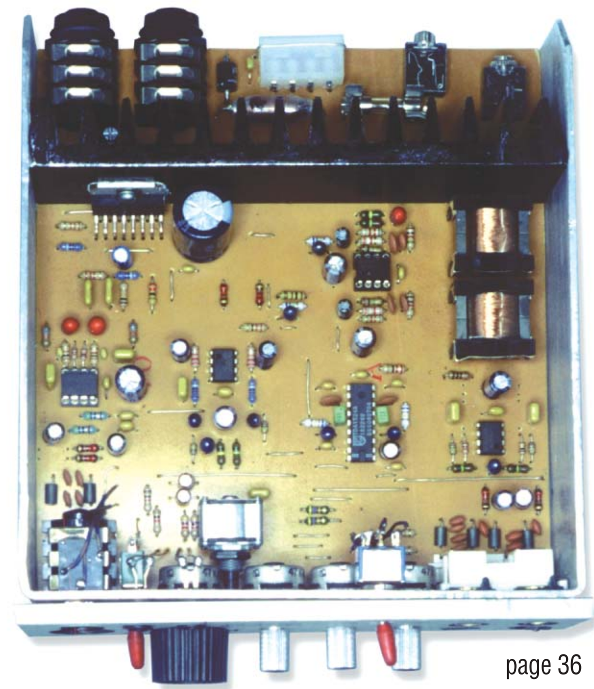

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The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it.

JOHN STUART MILL

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Remembering Ken Wilkinson

By Reg Williamson

Rarely can one describe someone as a legend in his or her own lifetime, but it would certainly apply to Decca's recording engineer, Ken Wilkinson. Affectionately known as "Wilkie," he was respected by his peers and, even more important, enjoyed the complete trust of the artists he served with his craft. Wilkie began his long career with a small company called World Echo, literally at the dawn of electrical recording in the mid-1920s, and in 1931 he moved to Decca joining the redoubtable Arthur Haddy. It was Haddy's team that developed the well-known Decca "Tree" for stereo, a T-shaped structure holding three omnidirectional Neumann microphones, which Wilkie used extensively.

It had often been said that you could recognize a Wilkie recording because of the rich balance, favoring to full measure the bottom end of the spectrum, and bringing out the acoustic of his favorite venues. He retired in 1980, just before digital recording began to displace analog. Paradoxically, his skills were recognized more in the US than in the UK, where he won the prestigious National Academy of Recorded Arts and Sciences Award twice; the first for Berlioz' "Symphonie Fantastique" in 1972 and Ravel's "Bolero" in 1977. He also worked for RCA and for Lyrita.

PHOTO 1: ***Left to Right:*** **Ken Wilkinson, Jimmy Walker (Producer for Lyrita).**

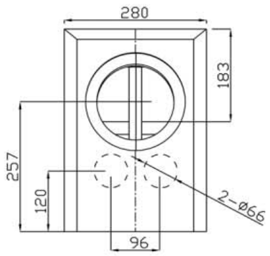
I spent a day with him about 14 years ago, recording his memories of the many famous artists with whom he had worked, and their idiosyncrasies. One will suffice. The conductor Stokowski was notorious for wanting to balance his recordings himself. Wilkie smiled, "Of course, maestro. I take it you won't mind if I come out and try my hand at conducting?"

Ken Wilkinson died at age 92 on January 30, 2004. ❖

Swans Tempus Kit

Swans proudly presents their first European-style, independently developed loudspeaker kit. Due to exceptionally high standards for a kit speaker, TEMPUS is a top-level, world-class performer in all regards.

German sound magazine KLANG+TON featured the Tempus in the June 2002 issue, commending its excellent acoustic response.



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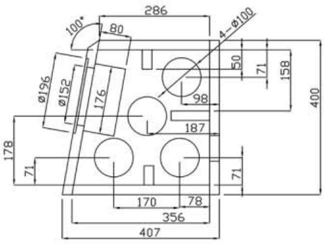

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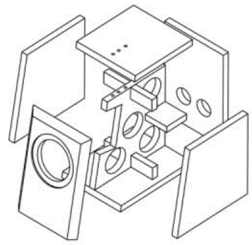

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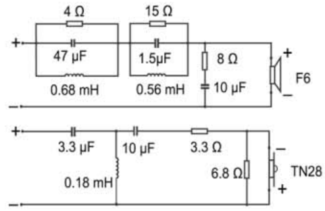

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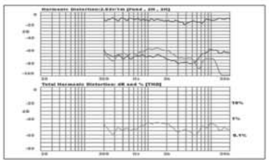

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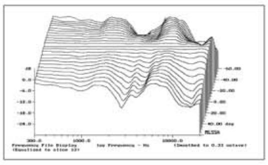

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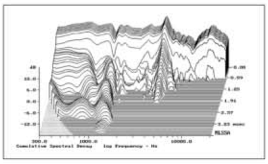

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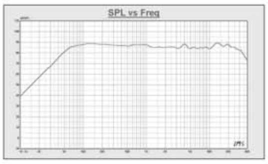

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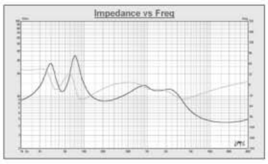

Tempus Impedance

The Tempus project was initially conceived as a private work of the acoustic arts, to be executed independent of all professional affiliations. Despite a lack of traditional marketing and initial editorial commentary, the design has since received wide acclaim in the independent press and has gone on to tremendous commercial success. Tempus' reputation is therefore solely the result of unexpectedly high fidelity from a speaker of its modest origin and cost.

For Tempus' development, Swans agreed to supply premium components for what evolved into a rigorous electro-acoustics design program, executed to the highest European standards. The resulting design's performance was subjected to stringent laboratory confirmation and completely documented in the audio press, including revealing measurements not typical for any loudspeaker much less one of this type. Tempus has since gone on to claim top performance awards from acclaimed KLANG+TON magazine of Germany, during which time Swans cancelled all advertisements and further commentary.

Tempus is now one of Europe's most favored loudspeakers, leaving listeners admiring its naturally musical sound and excellent value. The design is already a classic, destined to sweep the audio world in a fashion similar to its current success.

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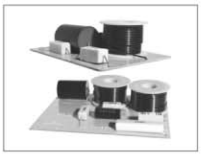

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Zen Variations 6: Son of Zen Gets Xploited

The Son of Zen amp grows up with the help of this simple Super-Symmetry circuit. **By Nelson Pass**

U.S. Patent #5,376,899 describes an amplifying circuit topology that takes advantage of the character of matched balanced amplifiers that are cross-coupled to provide cancellation of distortion and noise. The result provides high performance with very simple linear circuits and has been dubbed Super-Symmetry—an homage to particle physics—and is also known popularly as the X circuit. Super-Symmetry works by exploiting the complementary characteristics of matched balanced circuits to differentially reject distortion and noise, and applies a small amount of feedback to extend this symmetry, making the distortion and noise even more identical on each half of a balanced amplifying circuit.

In this article I am going to apply this technique to a previous project, the Son of Zen (*Audio Electronics* 2/97), which can also be found at www.passdiy.com and whose schematic is shown in *Fig. 1*. Here you see a very simple single-stage balanced differential amplifier without feedback or capacitors. Balanced input signal is presented to the + and – inputs, and a larger version of that balanced signal occurs across the output load.

WHAT IS SUPER-SYMMETRY?

To best understand Super-Symmetric feedback, consider *Fig. 2*, a simplified version of the Son of Zen (SOZ), where I arbitrarily inject a positive blip representing distortion or noise on the positive output. Because the SOZ amplifier has no feedback of any sort, there is no correction for this and it appears across the load.

Figure 3 shows the addition of two symmetric feedback loops to the circuit of *Fig. 2*, consisting of 1kΩ resistors in series with the inputs, and 10kΩ resistors connecting the input gates of the MOSFETs to the output drains, forming a local feedback loop for each transistor.

If you arbitrarily inject a blip of distortion into the output of Q2 as in *Fig. 2*, note that this blip will also appear (reduced in size) at the gate of Q2, and as a result will also appear as a blip of current coming out of the source of Q2. This current blip drives the source of Q1, and makes the blip appear on the drain of Q1, in phase with the blip on the drain of Q2. To the extent that the sizes and tim-

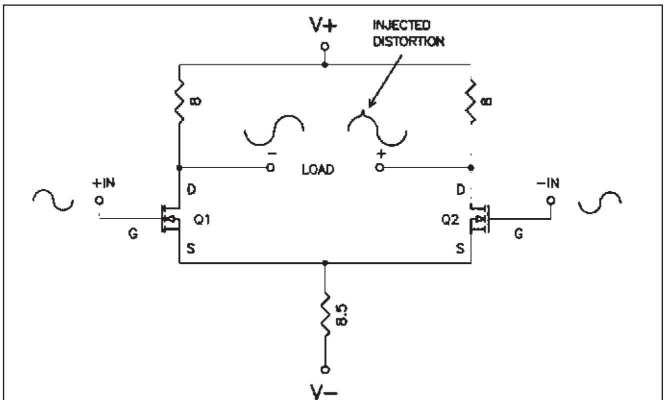

FIGURE 2: Distorting SOZ. A-2389-2

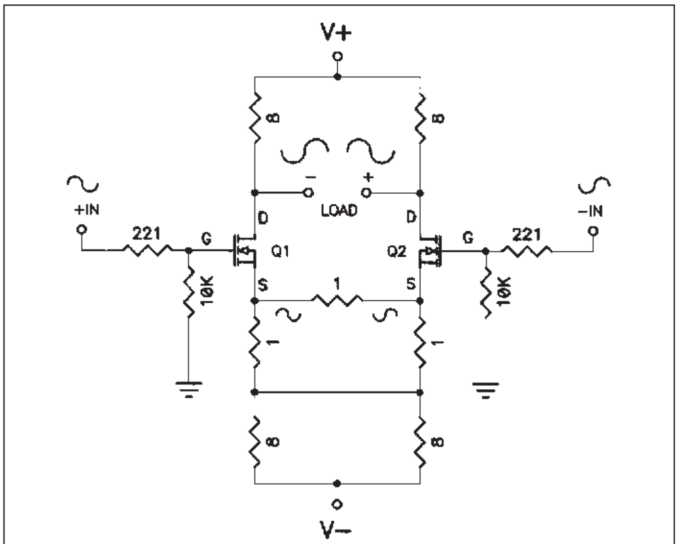

FIGURE 1: Original Son of Zen. A-2389-1

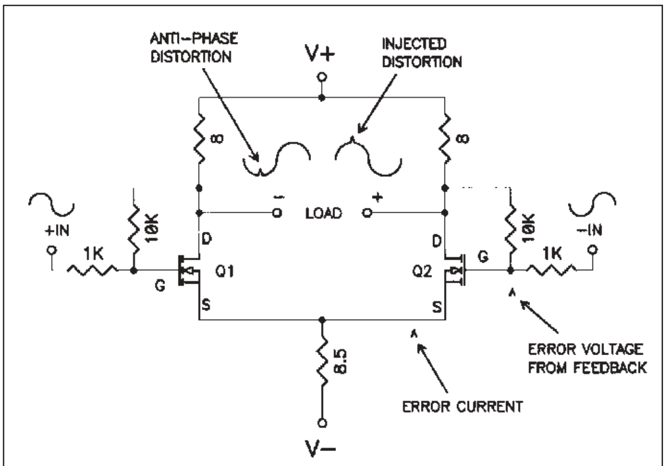

FIGURE 3: Symmetric correction. A-2389-3

ing of the two blips are identical, they cancel, and are not seen by the load, which is to say they disappear from the standpoint of the loudspeaker.

The unique thing about this arrangement is that the distortion contributed by each half of the amplifier appears nearly equally in phase at both outputs, but the signal appears out of phase at the outputs. So the desired signal is reinforced and the undesired noise and distortion is cancelled.

Super-Symmetry is ideally used to obtain high quality performance from very simple circuit topologies, avoiding the high order distortion character and feedback instabilities of complex circuits.

A single-stage circuit such as the SOZ would not actually be expected to see as much improvement as a two-stage circuit because, by its very nature, it does not correct for the nonlinearities in output current caused by the gate-to-source nonlinearities of the differential pair themselves. This is a fundamental limitation, and of course is true of differential pairs in general.

However, Super-Symmetry does correct the errors in the parts of the circuits that follow them. In a two- or three-gain stage design, the differential pair corrects the errors of the 2nd and 3rd stage, and, of course, any nonlinear-

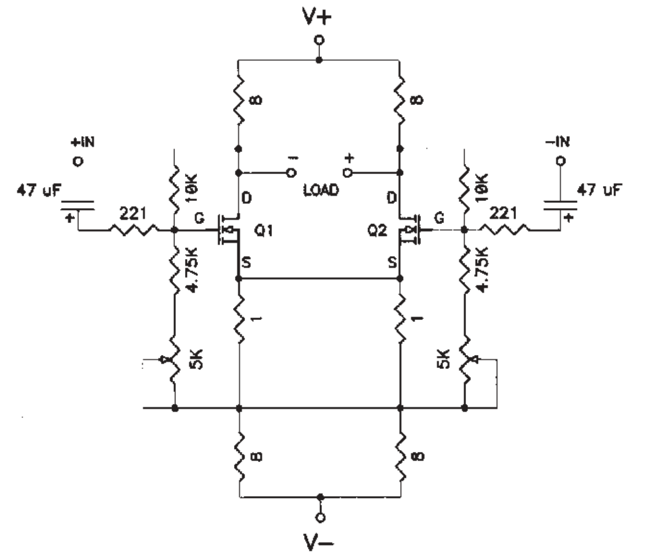

FIGURE 4: SuperSymmetric Son of Zen. A-2389-4

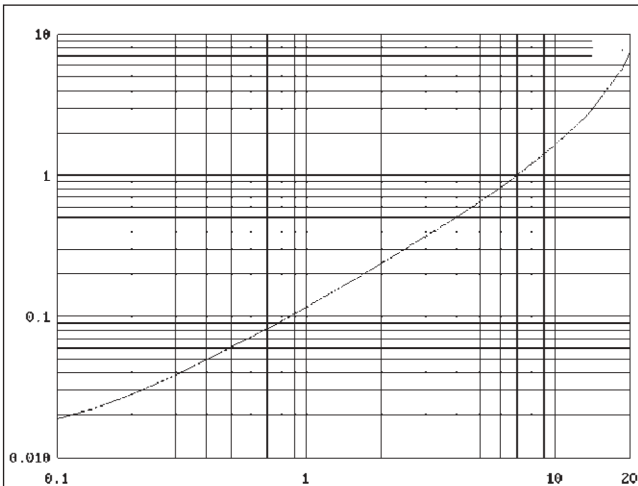

FIGURE 5: SOX amplitude measurement—30V. A-2389-5

ities in the output voltage contributed by the loudspeaker. Typically a two-gain stage version of this topology can outperform the three or more stages of a conventional amplifier. By gain stages, I generally mean how many transistors there are in series with the signal path, ignoring cascode transistors and counting complementary pairs as a single stage.

As a demonstration of the power of this approach, Pass Labs launched the first "X" amplifier, the X1000, as a 2-stage design with a small amount of feedback, delivering over 1kW with high performance. It has compared well against amplifiers having as many as nine sequential gain stages, and as of this writing is still in production.

The X feedback technique works less well with many gain stages and high open loop gains, mostly because the transit delays through complex circuits can cause a "hall of mirrors" effect, where the correction comes too late and the distortion bounces back and forth in the circuit. Ordinary feedback amplifiers suffer pretty much the same fate, unless they are deliberately slowed down by frequency compensation.

Why is this important? Because there is a different character of sound attributed to simple linear circuit topologies versus the complex circuits that obtain their performance through the generous use of feedback. It is difficult to produce high power amplifiers with simple circuits, and many previous efforts using many gain stages and high feedback have been described as powerful but musically clinical. Super-Symmetry makes it easier to produce high power amplifiers with the musical characteristic of simple low power amplifiers.

CONSTRUCTION

Figure 4 shows the actual circuit that I modified from the original SOZ. All the details of the original project apply to this amp, and I refer you to that article. I removed the 1Ω resistor between the sources of Q1 and Q2, and added two input capacitors of arbitrary value, 1kΩ input and 10kΩ feedback resistors, which set the gain of the circuit. Also, I used 4.75kΩ resistors in series with 5kΩ pots to set the operating point of the MOSFETs precisely for best performance. I will refer to this circuit hereafter as the SOX for brevity.

PERFORMANCE

In order to properly assess the differences between the original SOZ and one with the addition of symmetric feedback, I built a SOZ channel, made the appropriate measurements, and then removed the 1Ω source-to-source resistor and added the feedback elements to form the circuit of *Fig. 4*. To compare apples to apples, I used the

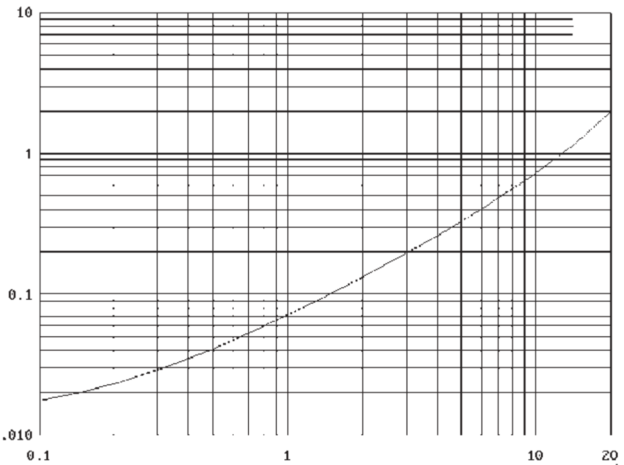

FIGURE 6: SOX amplitude measurement—40V. A-2389-6

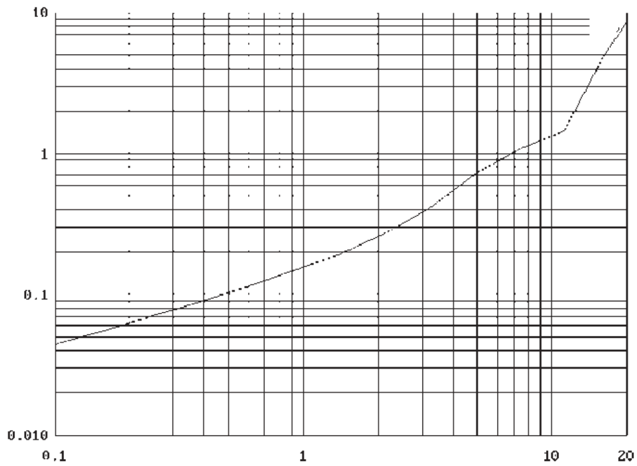

FIGURE 8: SOZ amplitude measurement—40V. A-2389-8

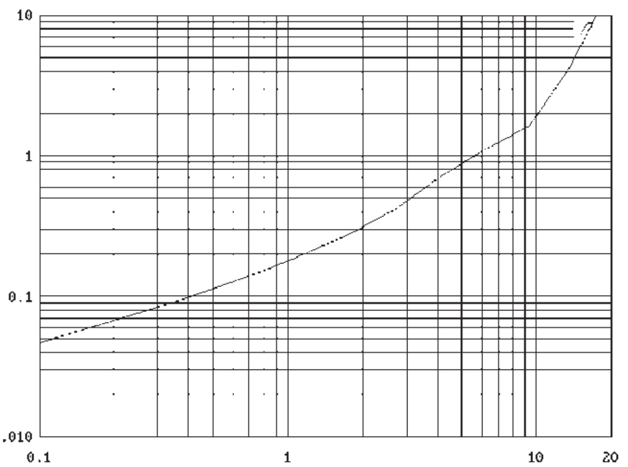

FIGURE 7: SOZ amplitude measurement—30V. A-2389-7

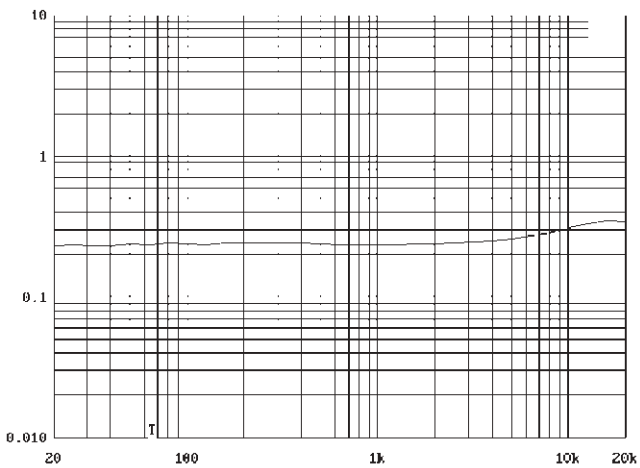

FIGURE 9: SOX frequency—balanced. A-2389-9

same supply, the same transistors, an identical layout and test setup, and designed the bias current and gain to be identical. The power supply consisted of a CLC network of 30,000µF, 2mH, and 30,000µF for each rail polarity. Again, you can find other details of the circuit in the original Son of Zen article.

A single stage is the best for tutorial examples, but is not otherwise the most ideal example of the power of Super-Symmetry. Nevertheless, I achieved some interesting and non-intuitive results.

Unless otherwise noted, the curves and data that follow were with an 8Ω resistive load, frequencies at 1kHz, output levels at 2W, 600Ω balanced source or 50Ω single-ended source, and either ±30V at 6A total bias or ±40V at 8A (6A bias means 3A through Q1 and Q2 each).

In the cases where the rails are at ±30V, I set the drain voltages of the SOX circuit on what I judged to be optimal at 8V referenced to ground, which was also about 12.5V drain-source. For the ±40V case, I set the drain voltage at 8.6V DC referenced to ground, which was 13.7V drain-source on the MOSFETs.

As previously noted, the SOZ and SOX gain figures were set about the same, which is close to 12dB with an 8Ω load. I set the offset potentiometers to attain the desired drain voltages with a difference of less than 50mV DC, which is then the DC offset experienced by the load. This should be readjusted after adequate warm-up, as the character of the MOSFETs drifts with temperature.

First off, I measured the open loop gain without feedback (and without the 1Ω resistor source-source) and achieved a figure of about 20dB. With a 12dB gain after the application of feedback, I calculated 8dB of feedback on the amplifier.

I measured the damping factor, which was about 0.5 (16Ω output impedance) for the SOZ and about 3 (2.7Ω output impedance) for the SOX—an improvement by a factor of 6 times—more than twice what we would expect with 8dB of feedback—a good example of the power of Super-Symmetry.

The input impedance of the SOZ will be whatever you set the input-ground resistors and the capacitance of the MOSFETs. After the addition of the feedback loops, the SOX input impedance measures approximately 4kΩ balanced and 2kΩ unbalanced. This figure is approximate, as it depends somewhat on the MOSFETs and the other settings. It's not a high figure, and it

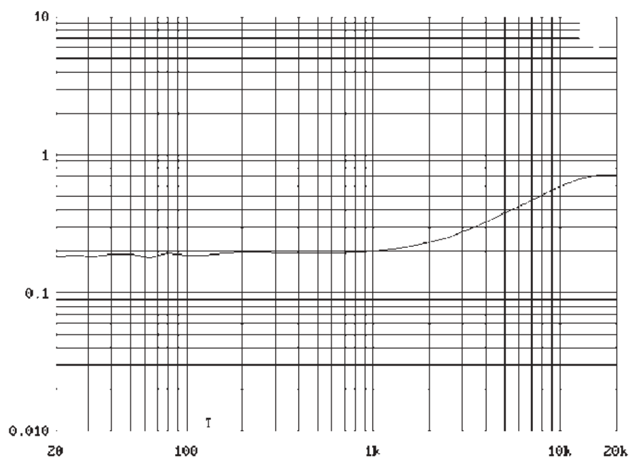

FIGURE 10: SOZ frequency—balanced. A-2389-10

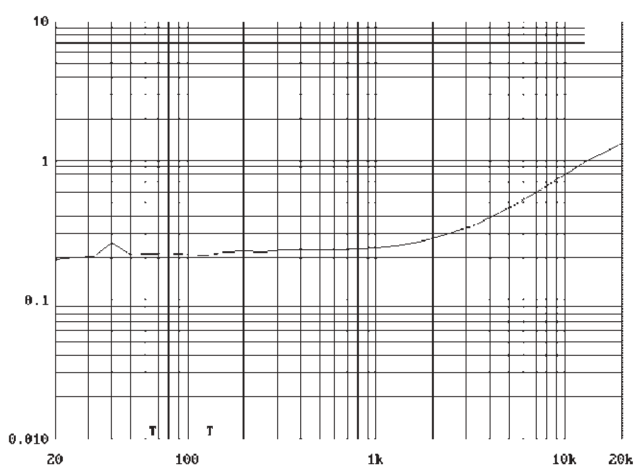

FIGURE 11: SOX frequency—single-ended. A-2389-11

can easily be raised to higher values by increasing the input network values proportionately, but at a cost of higher distortion at high frequencies. Alternatively, you can simply raise the value of the 1kΩ input resistor if you can afford to throw away some gain. In my experience, it will work well with any balanced source, and most single-ended sources, but there are some tube circuits out there that won't like it.

As before, there is no frequency compensation in the amplifier, and I have not encountered a case of instability over the years. It will drive anything—maybe not well, but without oscillation. You can short the outputs as you like, it will not damage the amp. I have not seen significant turn on/off thumps, and I don't expect you to. The amplifier benefits from Vgs matched transistors, but they are not required.

You can refer to the dissipation and power figures from the original SOZ article, and after the modification, the output power goes slightly higher for a given supply level, as there is no longer the loss across the 1Ω source-source resistor of the differential transistors, but otherwise the curves are very similar.

DISTORTION VERSUS AMPLITUDE AT 1KHZ, 8Ω

Figure 5 shows the distortion versus amplitude for the SOX circuit using 30V rails and a balanced 600Ω source. From the curves I would call this a 10W amplifier. *Figure 6* shows the same for 40V rails, and I would call this a 20W amplifier. For both cases, the overall efficiency is about 3%.

It's pretty clear that the higher voltage and bias of the latter gives marked improvement. In both cases you can see a small improvement over the SOZ, but nothing to write home to mom about. As pointed out earlier, I do not expect improvement in this regard at lower frequencies because for a single-stage amplifier, Super-Symmetric feedback does not correct the Vgs versus current distortion of the first differential pair, and that is mostly what you are measuring here.

DISTORTION VERSUS AMPLITUDE — SINGLE-ENDED SOURCE

The situation alters when the amplifier is driven from a single-ended source (and the minus input of the amplifier is grounded). The curves for the SOX amplifier do not change significantly, but you see that the distortion character for the original SOZ has roughly twice the distortion and begins clipping earlier. *Figure 7* shows the 30V supply example of the SOZ, and *Fig. 8* shows the 40V example.

DISTORTION VERSUS FREQUENCY — BALANCED SOURCE

For the SOX circuit you see a slight increase in distortion (about twice) as you approach 20kHz at 2W (*Fig. 9*), and for the SOZ you see a more dramatic climb

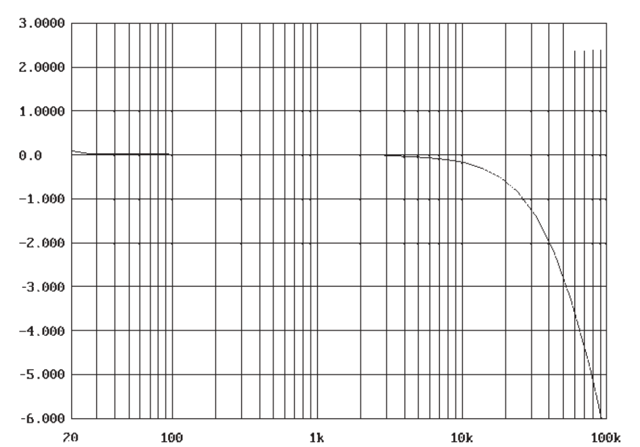

FIGURE 12: SOZ frequency response. A-2389-12

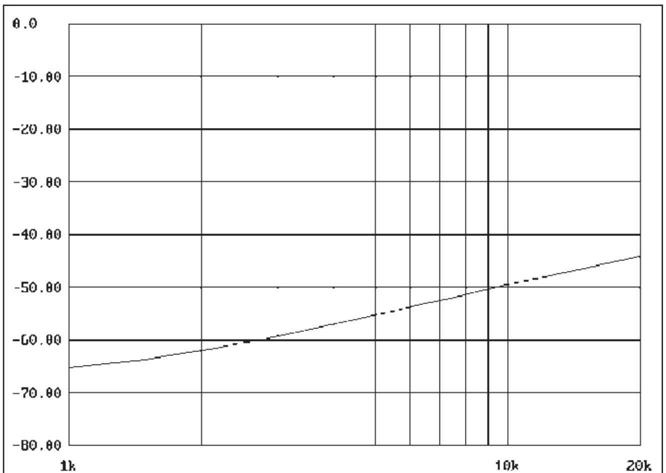

FIGURE 14: SOZ common-mode rejection. A-2389-14

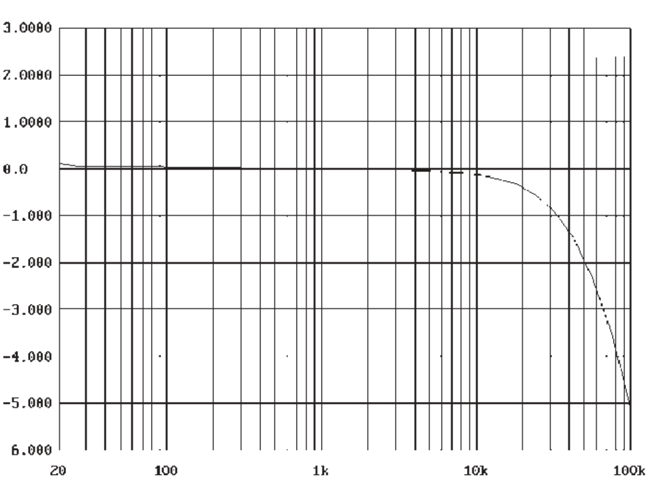

FIGURE 13: SOX frequency response. A-2389-13

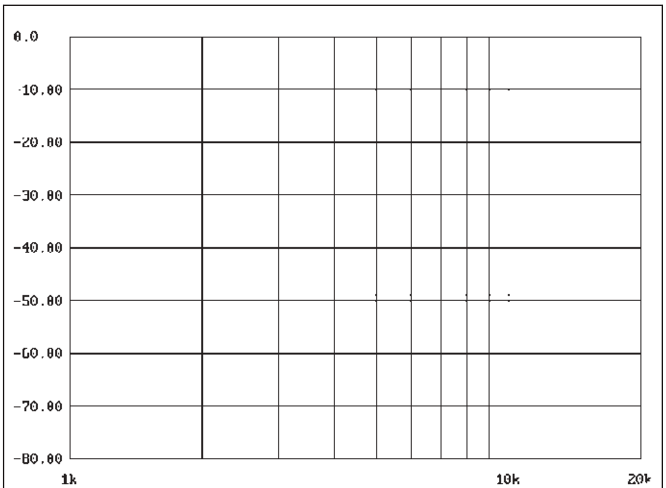

FIGURE 15: SOX common-mode rejection. A-2389-15

(*Fig. 10*). I have not explored this difference thoroughly, but I am assuming that the feedback in SOX is removing device distortion outside of the Vgs/current effect, most likely the nonlinearity in capacitances between the drain-source and drain-gate.

DISTORTION VERSUS FREQUENCY — SINGLE-ENDED SOURCE

Figure 11 shows the increases that occur in distortion versus frequency in SOX when driven by a single-ended source. It goes up quite a bit toward 20kHz, but less than half that of SOZ. You can use balanced sources here, or you can console yourselves in the fact that musical material declines in energy above 5kHz.

FREQUENCY RESPONSE

The high-frequency rolloffs for both SOZ and SOX do not alter appreciably for different supply voltages and bias currents, but the SOX does come out faster, with a –3dB point at 70kHz versus 50kHz on the SOZ (*Figs. 12* and *13*). These figures are unaffected by the use of balanced versus unbalanced inputs.

COMMON-MODE REJECTION

Both circuits show excellent input common-mode rejection (CMRR), the ability to ignore noise appearing the same on both balanced inputs. The SOZ was slightly better (–65dB) at low frequencies, but declined at high frequencies, while the SOX was about –50dB, but constant across the audio band (*Figs. 14* and *15*).

LISTENING TESTS

When your output power is 20W with an efficiency of 3%, you tend toward the more efficient type of loudspeakers. Fine, that's what I prefer anyway. I spent some quality time with the SOX and several drivers mounted in the J-Low rear-loaded horns ("The J-Low Project," *audioXpress* Feb. 2004), also at www.passdiy.com. The drivers were the Jordan 92, Manger, Fostex FE208 Sigma, and the Lowther DX55 with the "shower head" phase plug. The Manger and Jordan are not that efficient compared to the Fostex and Lowther, but all are interesting examples of single full-range drivers, particularly when augmented by a bass horn.

I don't want to get into the relative merits of the drivers here (I will elsewhere), but I did compare these to the stock SOZ (actually, the same amps, reconfigured), a pair of balanced Zen Lite (with the three bulbs), and the Aleph 30.

Previously, I had preferred the Aleph 30 on all the drivers except the Jordan, where the Zen Lite ("Zen Variations Part 1," *audioXpress*, March 2002) sounded a little more musical. Part of this is because the other drivers, in spite of their varying efficiency, have larger cone areas and can make more use of the greater power of the Aleph 30. Previously the SOZ sat somewhere in between the other two amps. I rather expected to hear improvement in bottom end control and a more even spectral balance from the SOX. Maybe it's a self-fulfilling prophecy, but that's pretty much what I received.

Compared to the SOZ, there is definitely cleaner bottom and top, and more ability with complex music, particularly orchestral. If it was just a piano or vocal alone, the difference was not as great. I heard a bit better spectral balance in general, and the result was more musical and toe-tapping.

The most interesting experience, where the amplifier seemed outstanding, was with the Lowther on piano, material with dominant vocals and not much else, or string quartets (Philip Glass recordings, as an example). You could kill an evening pretty easily with this if you didn't try to play it too loud or expect deep bottom end.

Compared to the Zen Lite, it depended on the bias conditions. I used a Variac to go between the two 30V and 40V settings, and at the higher bias, it definitely was better than Zen Lite and approached the Aleph 30 for dynamic range, and I would say matched it overall on most material. Unfortunately, at 40V rails and 4A bias per transistor (640W/ch dissipation), it ran hot as hell, and my heatsinking was not up to the job over long periods. As a result, quite a bit of the time was spent at the lower settings.

Alas, the lower setting is slightly inferior, and it's quite easy to get spoiled. This was true of the original SOZ as well, so some things just don't change.

An argument is often made that amplifiers like this are wasteful of energy and not environmentally friendly. Well, that's true, but I get more enjoyment out of an amplifier like this than driving a car, which consumes between 10 and 100 times more power. According to my electric bill, for the couple hours I might spend listening each day, an amp like this represents about 2% of my overall consumption. I think I would not leave it on 24 hours a day, though.

To sum up, I am very happy with the results here. If you have already built a Son of Zen, then I certainly recommend that you run out and buy about 10 dollars' worth of parts and upgrade it. If you haven't already built a Son of Zen, stay tuned, and we'll see about raising that efficiency figure real soon.

Nelson Pass, the proprietor of Pass Laboratories and a frequent contributor to *audioXpress*, enjoys creating trouble.

Tube Audio Construction Tips: Part 2: Sockets

This section on tube sockets shows you how you can enjoy audio construction and still keep your bankroll afloat.

By Graham Dicker and Jim Tregellas

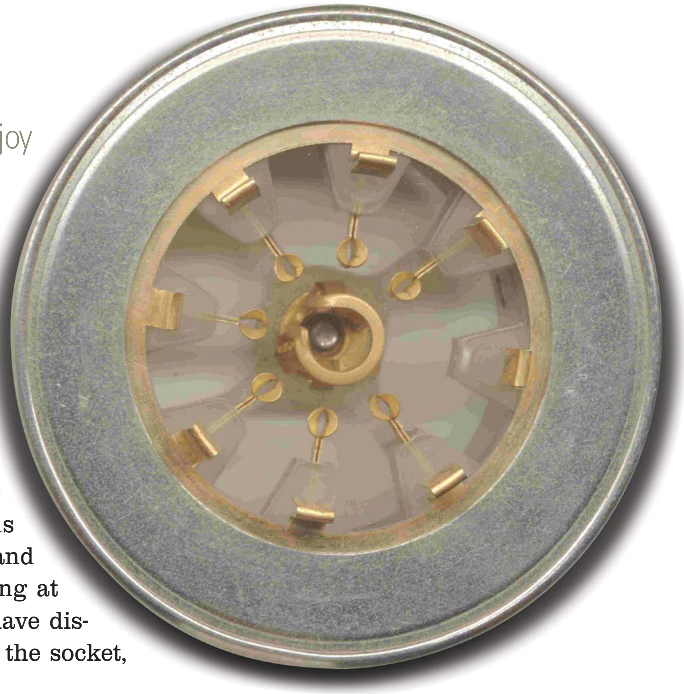

PHOTO 1: SK600 Eimac air socket.

In part one of this series, I covered the purchase of tubes for audio projects, and explained my particular interest in using transmitting tubes for SE amplifiers. While the tubes are available at a reasonable price, the sockets represent another problem that can quickly empty a healthy bank balance. Help is available. Here are some ideas I have used in the past to reduce costs.

TUBE SOCKETS

You can scrounge octal, noval, and seven-pin tube sockets from old black-and-white TV sets. There are many suppliers of new and NOS sockets on the web and advertised in *audioXpress*. The better sockets have ceramic substrates. Some vintage pre-WWII tubes require wafer-type sockets. If you purchase these, ensure that there are no heat marks or carbon tracing in the wafers.

The second most common failure with older sockets is tarnishing (oxidation) and corrosion of the socket pins. High resistance on filament pins can shorten tube life and cause excessive heating at the tube base. If you have discolored heater pins on the socket, beware.

If you have oxidation or tarnishing on tube sockets or NOS components, you can often remove this by soaking the socket or part in a solution of citric acid, which is available at the supermarket. Citric acid, which is used in cooking, is not a strong enough acid to eat away the metal parts, but it does act as a good descaler. After removing the oxidation, I always soak the socket in hot soapy water to remove all traces of the acid.

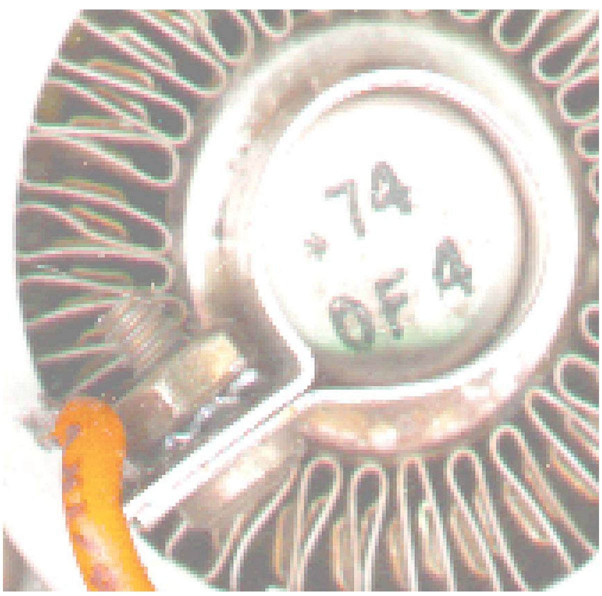

PHOTO 2: Homemade anode clamp.

You can also clean used sockets of flux and surface scale with an old toothbrush and some acetone (paint thinner). You can restore aluminum IF cans by using a wire brush and steel wool, and then anodizing them in a weak solution of sulfuric acid, or caustic soda. Always use rubber gloves and much care when handling any chemical.

When I first started building low-cost amplifiers, I found that one of the most expensive items was the tube socket for transmitting tubes. You will usually pay around $50 to $100 per tube socket for new ones. For the ceramic tubes, especially the 4CX150, 4CX250B, and 4CX350A, you do not need to purchase an expensive air socket. My handy hint here is that you can purchase a standard Locktal socket for around $5. Be sure to purchase the ones without the mounting shrouds, so as not to impede the airflow over the tube base.

For your audio purposes, you do not need the expensive ones with built-in screen bypass capacitors that sell for $100. Also, do not purchase the ones intended for grounded grid operation with an internal grid bypass capacitor (*Photo 1*), unless you wish to build a grounded grid audio amplifier. Many years ago, I used a pair of these grounded grid sockets by mistake and had a massive rolloff of high frequencies and poor slew rate in the output stage that drove me nuts for weeks, until the penny dropped. What made life worse was the 15dB of feedback around the stage that was doing really weird things. I tried everything, including a new output transformer.

You can easily form an anode clamp

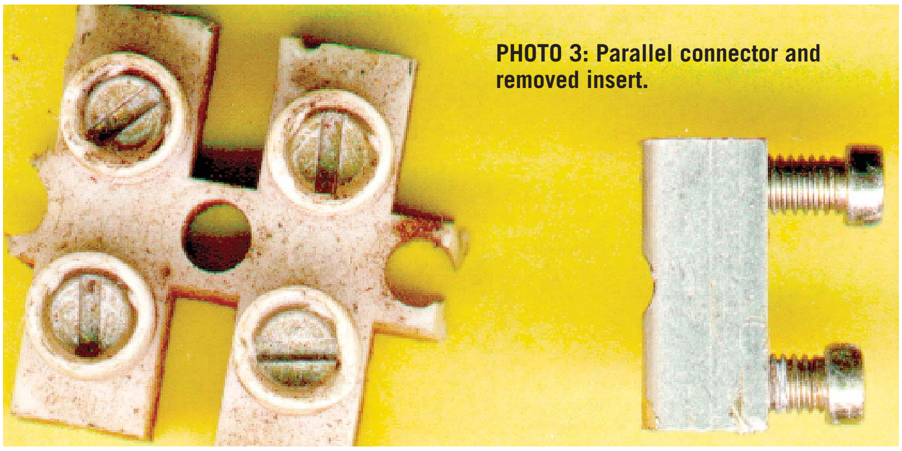

PHOTO 3: Parallel connector and removed insert.

PHOTO 4: Socket pin inserts made on a lathe.

from a short length of aluminum wrapped around a drill bit (*Photo 2*), or you can purchase a radiator hose clamp with equal results. For tubes such as the GI-7B ceramic triode, you can use hose clamps for all of the tube base connectors. I have also used recycled can-type electrolytic capacitor clamps.

MAKE YOUR OWN

If you are on a really tight budget, why purchase a base at all, when you can make your own? In my cheapskate mode of playing with devices that glow in the dark (tubes), I have often needed a tube base that I did not have in my junk box.

A good trick here is that you can mechanically support your tube using a radiator clamp, then connect to the tube pins using individual Molex crimp connectors. There are many different sizes available. Use the right diameter ones to slip over the tube pins. There is enough grip with these so that they do not fall off. If you wish, you can mount these on a piece of Bakelite or fiberglass substrate by drilling appropriately spaced holes and using ten-minute epoxy to secure them.

For glass tubes such as the 813 or GK–17, the RCA 4–125A, and the Eimac 304TL and others, I make my own sockets by purchasing a ceramic bathroom tile, drilling it and mounting parallel connector inserts for the pins (*Photo 3*). Ceramic tiles come in different sorts, textures, colors, and characteristics. I use wall tiles because they are easier to drill than harder floor tiles, and because of the heat resistant properties, not mechanical strength. A single wall tile will yield around 12 ceramic tube sockets. At a cost of around $1 per tile, that's less than ten cents each socket.

You can cut a standard tile with a tile cutter or a diamond scriber, and then

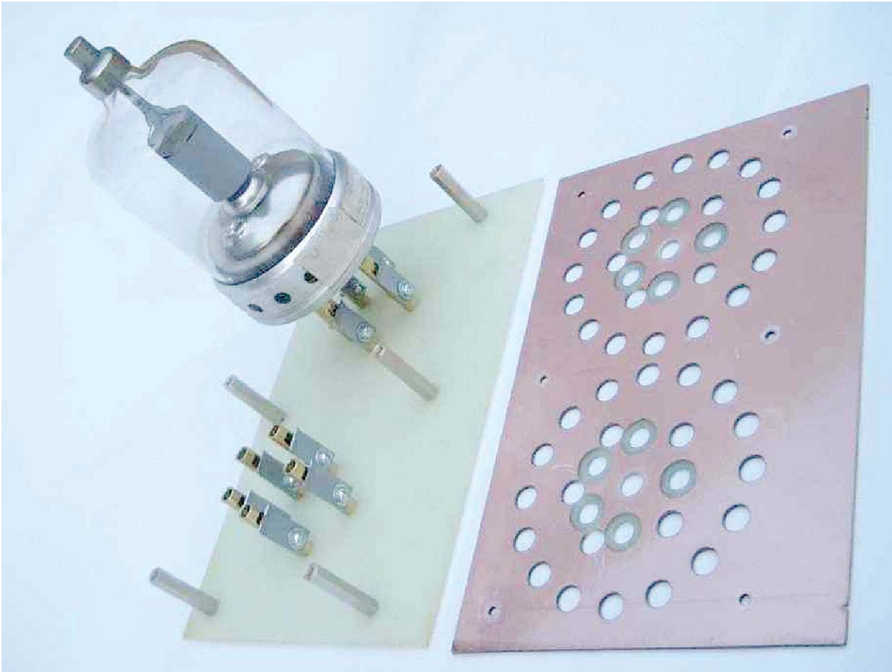

PHOTO 5: Completed 4–125A socket and top ground plane.

sand the sides to remove any edges. The best method that I have found to drill holes is to use a section of masking tape over the area where the hole is to be drilled, and then use a masonry drill bit on low speed. You can easily fabricate a template from some clear Perspex.

Make the connections with Molex crimp connectors, or for the larger tubes, I use brass inserts removed from parallel connectors. These are available in a range of sizes to suit most tube pins. The added advantage is that if the connections are screwed, the tubes do not fall out of the sockets.

The automotive industry makes a range of suitable push-on connectors and bullets which you can also use. Brass connectors are recommended for filament connectors due to the high current requirements. The added advantage is that brass connectors have less oxidation problems over the life of the connector.

Having a good workshop with a lathe makes other solutions possible. As *Photo 4* shows, two possible connector forms evolved, both made from 0.250″ diameter brass rod. The first version (shown on the right) simply involved drilling a 4.5mm diameter hole for the tube pin and a 2.5mm hole for the 3mm screw on either end of the rod. I then "necked" the rod to reduce the wall thickness around the pin, splitting the necked section horizontally into four parts with the lathe chuck stationary and with a very sharp V tool. I mounted the tool in the tool holder, so that it cut at 90° to its normal direction (longitudinally rather than vertically). I then gently squeezed these sections on the finished socket (*Photo 5*) inwards using pliers to give a good springy grasp of the tube pin.

The second connector form on the left is more difficult to make, but gives an extremely firm grasp on the pin. I made the connector by drilling a 4.5mm hole for the tube pin and two 2.5mm holes for the screws, and then milling a double height flat on the socket using a tungsten carbide rotary file in a drill press. I fabricated the spring from 316 stainless steel. Socket pressures are such that the 4–125 filament current presents no problem with this connector. I ended up using this as my preferred socket.

I then mounted the connectors between two pieces of printed circuit board. I drilled the upper one to provide airflow around the tube and to locate the upper parts of the sockets, and drilled the lower one simply to accommodate the sockets and inter PCB spacers. Except for an insulating area around the sockets, I left the copper on the upper PCB, and removed all copper from the lower PCB. I have a very nice custom-made ground plane tube socket for 4–125s complete with built-in airflow direction (*Photo 5*).

Having made these sockets, I suggest that brass connectors from trailer light plugs and sockets can work equally well. While it was a long task, the first ceramic socket I ever made used an entire marble color tile as the top of a chassis for the whole amplifier. It looked great, but took ages to drill all of the mounting holes. If you have the budget, I recommend getting them hydra- or laser-cut.

CHEAP FANS AND COOLING

One of the lowest-cost blowers I have in my workshop for using with ceramic tubes is an old hairdryer, which makes a great centrifugal blower for the 4CX series tubes. I use a short length of Teflon held in place with a rubber band to connect the two together. You can easily make a simple chimney from a small piece of Teflon sheeting rolled into a tube and secured with a rubber band or a radiator hose clamp.

A blown cavity can be fabricated using PVC plumbing fittings and a recycled 36′/minute Panaflow computer fan with equal success. You can also use these fans to cool glass tubes. I prefer to use fans in pairs: one blowing cool air onto the tube, the second to suck out the hot air from the other side. An alternative chimney for tubes is a small ceramic garden pot with holes drilled in the bottom, then placed over the tube to direct a cool airflow. ❖

Vented 8″ Driver Subwoofer, Part 1

This article covers the development and construction of physically small subwoofer systems using 8″ drivers.

By G.R. Koonce and R.O. Wright, Jr.

What is an apartment subwoofer? We are talking about small subwoofers that might not get your lease terminated! While they may not play as loudly as larger drivers, they do provide true subwoofer performance. The designs use single voice coil drivers and are intended for use in pairs.

A DIFFERENT DESIGN APPROACH

You would think development of an 8″ driver subwoofer would be as easy as building with a larger driver. This turned out not to be true with the 8″ drivers we located.

After development of good 12″ and 15″ driver subwoofers[1] using the Infinite Box (IB) approach[2], we had hoped to go the same route with the 8″ drivers. While the IB approach provides great-sounding speakers in small enclosures, it produces a higher system –3dB frequency (f_3) than some other approaches. While measurement showed the 12″ IB subwoofer went down to 25Hz, the breadboard 8″ driver IBs yielded f_3 limits in the 50 to 80Hz range—clearly not acceptable. The IB approach fails here because of the relatively high f_s of the 8″ subwoofer drivers. Since a given driver will show a similar high f_3 in a closed box, they are not a viable alternate.

The approaches left to achieve a low f_3 while retaining a small enclosure are the passive radiator (PR) and the vented box (VB). If you can obtain small PR units which provide sufficient displacement, the PR technique will enable a smaller enclosure. This is because adding one or more PR units will require less enclosure volume than the duct of the VB.

With a VB design the port delivers most of the system output at low frequency and thus must pass a large air volume. If you decide that a 4″ ID port is sufficient, which may be optimistic, you will discover the required port duct length is in the 40″ to 60″ range. No matter how you coil that up it takes a lot of volume, close to $\frac{1}{2}ft^3$.

We were aware that Al Wooley (of RAW Acoustics) and Bob Reimer (of Creative Sound Solutions) were developing a small PR subwoofer using an 8″ driver from TBspeakers. We thus decided to pursue the VB approach and try to solve the "huge vent duct" problem. We hope to publish information on construction and performance of the PR subwoofer in a future issue.

HOW BIG A VENT DUCT?

Recently published works on the requirements for high-

PHOTO 1: Front view of MCM #55-2421 (#194) driver.

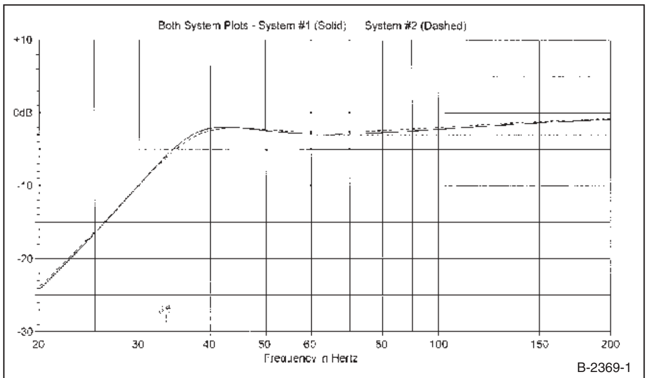

FIGURE 1: Computer VB designs for MCM #55-2421 (#194) driver pair.

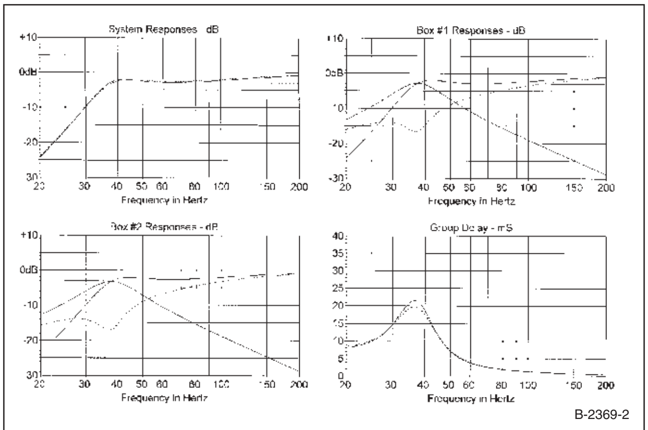

FIGURE 2: Other system performance for MCM #55-2421 (#194) driver pair.

power vent ducts have not been encouraging. Reference 3 looks at the problem in terms of Reynolds numbers. Testing showed that for a 100W power level even a 6″ ID vent duct was not sufficient to prevent response shape change with power level. A 100W steady-state input power level is a more severe requirement than our intended application.

Reference 4 is a detailed study of duct shape. It included a chart by Harwood that related maximum SPL to box tuned frequency (f_B) for various duct sizes. In our case the #197 driver (identified later) is rated at 84dB/W/m with a 120W power capability, so it could develop 105dB SPL at 1m. From the Harwood chart, at 30Hz, this would require a duct of about 140mm (5.51″) diameter.

Reference 5 studied the advantages of having flares at both ends of the duct. It indicated that such flares improved performance by reducing port-generated noise and allowing a given diameter port to be used to a higher power level before it "packed up." This article also pointed out that Precision Sound Products offers kits of the Precision Port™ that contain a flare for each end and a length of pipe you can cut to length. The kits are available in 3″ and 4″ ID sizes.

Taking a more direct approach, ROW contacted some of his pro-sound field acquaintances. They regularly work the problem of high-power VB designs used at low frequency, but generally with larger drivers. They reported the following rule-of-thumb for sizing a straight port: If the port diameter is 40% to 50% of the cone diameter you will be OK. David Weems[6] discovered a rule-of-thumb that states the port area should be between 1⁄7 and 2⁄3 of the cone area. Converted to a percentage of effective cone diameter, this is 38% to 82%, the minimum being in general agreement with the pro-sound value.

Our drivers have about a 6.5″ effective cone diameter, so the 40% to 50% rule indicates a port of from 2.6″ to 3.25″ diameter would be OK. We worry

PHOTO 2: Back view of MCM #55-2421 (#194) driver.

PHOTO 3: Front view of MA-Audio MA8OXE (#196) driver.

that these rules-of-thumb may have been developed for woofer and not subwoofer drivers. For a given cone size the subwoofer driver will move a lot more air due to its much larger X_{MAX}. We decided to design with a port area equivalent to a straight 4″ ID pipe (12.57 in^2) with a fallback to using the area of a 3″ ID pipe (7.09 in^2) with flares added at both ends.

DRIVER CANDIDATES

We had located and obtained pairs of the following driver types:

1. MCM #55-2421 8″—Our reference #194. See *Table 1* for what catalog data we have on this driver. *Photos 1* and *2* show this driver.
2. MA-Audio MA80XE 8″—Our reference #196. See *Table 1* for catalog data and *Photos 3*, *4*, and *5*. This driver is intended for automotive application and has the openings to the rear of the cone almost vertical to the rim (*Photo 5*). This driver would require major relieving of a thick front panel to allow the rear of the cone to breathe properly. The MA80XE has dual 4Ω voice coils.
3. TBspeakers W8-704C 8″—Our reference #197. See *Table 1* for catalog data and *Photos 6* and *7*. Unit A is serial No. 59 and unit B is serial No. 58.

We ran these drivers at low frequency for over 40 hours at a power level of 2–4W. As noted, we tested the drivers in a small IB breadboard and then measured the T/S parameters. These drivers have large vented magnet structures, making the volume taken from a test box hard to establish, so you should consider the test V_{AS} values reported as approximations. The #196 (MA Audio) driver had shown poorly in designs based on catalog data and had a shape making volume estimation very difficult, so we measured only a single unit.

Table 2 compares catalog and measured data for the #194 (MCM) driver. The two units are fairly consistent; however, they don't match the catalog data very well. First, f_S is much higher. These are foam surround drivers that might exhibit a drop in fs with additional use. The electrical Q (and thus Q_{TS}) is also much higher than specified, which is probably good for a VB application.

Generally the most variable parameter of a driver is the mechanical Q (Q_{MS}), a measure of the losses in the suspensions. Surprisingly, this is the parameter that agrees best with the specifications for this driver.

Table 3 shows the test results of one #196 (MA Audio) driver with the two voice coils connected in series. The catalog rated f_S (38Hz) was already rather

TABLE 1
DATA FOR 8″ DRIVERS OBTAINED FOR SUBWOOFER CONSIDERATION

Reference number	194	196	197
Make	MCM	MA-Audio	TBspeakers
Driver	#55-2421	MA80XE	W8-704C
Nominal Z_{IN} (ohms)	4	Dual 4	4
DCR (ohms)	3.4		3.2
Sensitivity (dB/W/M)	87	87.3	84
Rated Power (watts)	120	250	120
Maximum Power (watts)	240	500	230
X_{MAX} (mm — one way)*	16	10	12
Resonance — f_S (Hz)	24.7	38.3	28
Total Q — Q_{TS}	0.18	0.31	0.30
Compliance — V_{AS} (ft^3)	1.39	0.753	0.812
VC Diameter (mm)			50
Force Factor, BL (TM)	13.46		13.35
Magnet Weight (Oz)	47		50
Magnet gap height (mm)	8		8
Moving Mass (G)	64.2		95
Sus. Compliance (M/N)			333
Eff. Piston Diam.(inches)	6.41		6.59

*We have some question on these Xmax values. The MA80XE specified 20mm for peak-to-peak so 10mm one-way is valid. The other two drivers did not state how Xmax was defined or determined.

TABLE 2
CATALOG VERSUS MEASURED DATA FOR MCM #55-2421 (#194) DRIVER

	CATALOG	UNIT A	UNIT B
f_S — Hz	24.7	31.6	30.5
Q_{ES}	0.19	0.263	0.265
Q_{MS}	12.64	11.98	13.04
Q_{TS}	0.18	0.258	0.260
V_{AS} — ft^3	1.39	0.70	0.85
V_{AS} — L	39.4	19.9	24.2
SPL dB/W/m	87	85.9	85.9

TABLE 3
CATALOG VERSUS MEASURED DATA FOR MA-AUDIO MA80XE (#196) DRIVER

	CATALOG	UNIT A
f_S — Hz	38.3	46.9
Q_{ES}	0.34	0.47
Q_{MS}	4.23	5.88
Q_{TS}	0.31	0.44
V_{AS} — ft^3	0.75	0.41
V_{AS} — L	21.3	11.7
SPL dB/W/m	87.3	85.9

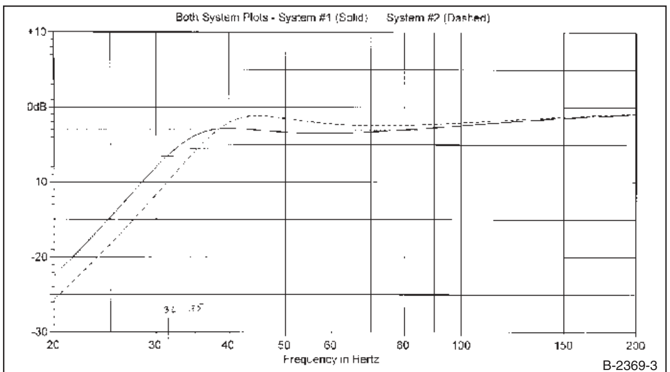

FIGURE 3: Effects of changing the tuning with MCM driver unit A.

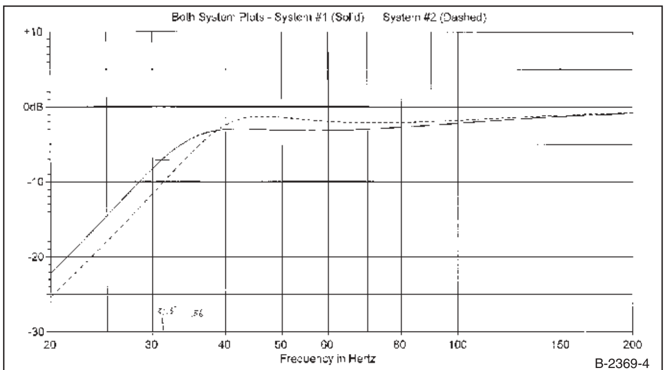

FIGURE 4: Effects of changing the tuning with MCM driver unit B.

high, but the real f_S (47Hz) was much too high for a subwoofer application. Again, this is a foam surround driver, and f_S may drop a bit more with additional use. Based on the high f_S, we deleted this driver from further consideration in a small subwoofer application.

Table 4 shows the test results for the #197 (TBspeakers) driver. The two samples match pretty well and are not far from the catalog specification. The low f_S values agree well with the catalog value and explain why this driver type was the closest to making it in an IB enclosure.

The V_{AS} measured a bit lower than specified. The electrical Q (and thus Q_{TS}) is considerably higher than specified, which, again, is probably good for a VB application. The SPL, which was already low, computed very low. This testing drop in SPL may be partly due to the lower V_{AS} (which may be partially a test problem) and partly to a "weaker" than intended motor as evidenced by the higher electrical Q.

SUBWOOFER VENTED BOX DESIGNS

Designing a VB for a subwoofer is a bit different from that for a normal woofer application. First, there is no reason to use a "flat" response alignment. The subwoofer is going to be used only from f_3 up to about 100Hz. This opens up

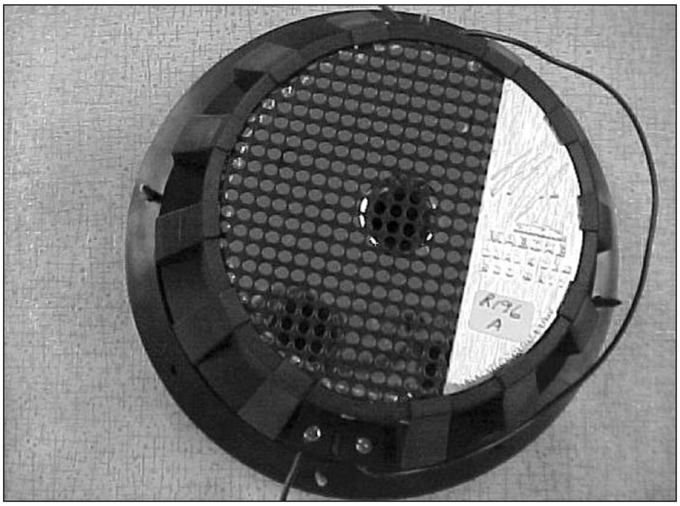

PHOTO 4: Back view of MA-Audio MA8OXE (#196) driver.

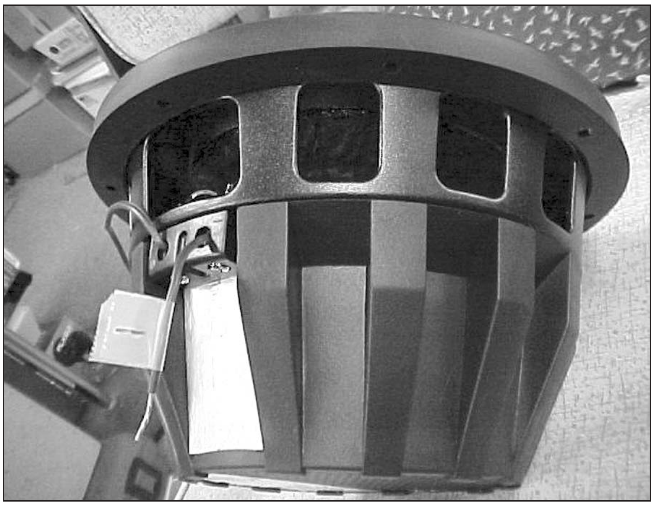

PHOTO 5: View of MA-Audio MA8OXE (#196) driver showing near vertical side openings.

alignment possibilities that might not be satisfactory for a woofer design.

We had already established that the vent duct volume was the major problem with the small driver VB subwoofer. For a given duct diameter, the duct length increases slowly as the box size decreases and increases quickly as f_B (box tuning) decreases. To minimize the duct length, use the largest possible box and the highest possible f_B. These are not requirements that help you produce a small subwoofer enclosure with a low f_3 cutoff frequency!

We investigated the MCM #55-2421 (#194) and TBspeakers W8-704C (#197) drivers via computer to establish designs of acceptable size and cutoff frequency that had reasonable duct lengths.

TABLE 4
CATALOG VERSUS MEASURED DATA FOR TBspeakers W8-704C (#197) DRIVER

	CATALOG	UNIT A	UNIT B
f_S — Hz	28	29.8	29.7
Q_{ES}	0.30	0.52	0.47
Q_{MS}	10.33	11.86	11.78
Q_{TS}	0.30	0.50	0.45
V_{AS} — ft^3	0.81	0.61	0.61
V_{AS} — L	23	17.1	17.2
SPL dB/W/m	84	81.2	81.6

MCM DRIVER

The alignment developed for this driver has a bass response that is "shelved" down from the passband response. While this results in some sensitivity loss, the intent was to keep box volume (V_B) and f_B as high as possible to minimize vent duct length. *Figure 1* shows performance of the VB design for the pair of drivers. The net box volume is $0.7ft^3$ and the f_B is about 37.5Hz.

The performance of the two samples match well in this box containing absorption damping; note Q_A is set to 10 for a total box Q of 4.8. This alignment's shelved LF response pushes f_3 down to about 34Hz for a loss in sensitivity of perhaps 3dB.

Figure 2 shows other system results. Note that the port output dominates the system output below 52Hz and the group delay curve looks OK for a subwoofer. *Figures 3* and *4* show the effect on the two samples of altering the tuning by ±2.5Hz. It is clear you can play with the tuning to trade off sensitivity loss for a lower f_3. Using equations provided by David Weems, based on years of studying duct length in boxes containing damping, the computed length for f_B = 38Hz is about 29″ for a 4″ ID vent duct and about 15½″ for a 3″ ID vent duct.

TBSPEAKERS DRIVER

The design software showed that changing net V_B over the range 0.5 to $0.9ft^3$ had almost no effect on the highest f_B value that could be used with the #197 driver. Since duct length changes rapidly with f_B and slowly with V_B, this was not a great help. Again, the designs developed were for damped boxes with Q_A = 10, total box Q = 4.8.

We liked the results best with a net V_B of $0.8ft^3$ and f_B around 32Hz. *Figure 5* shows both driver samples in this design yielding f_3 around 30Hz. Note there is no sensitivity loss with this design, the response showing a dB or two

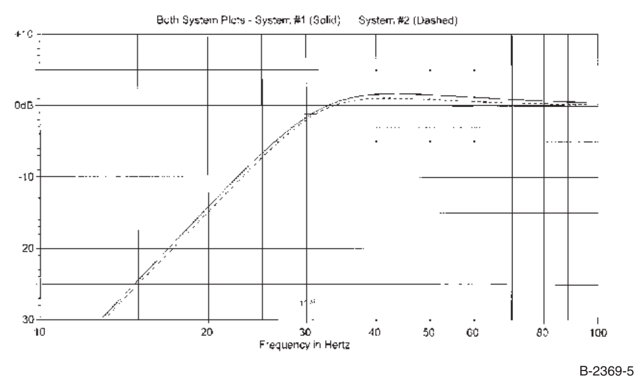

FIGURE 5: Computer VB designs for TBspeakers W8-704C (#197) driver pair.

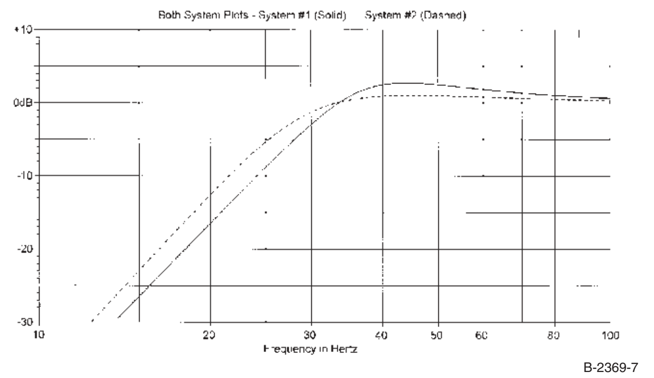

FIGURE 7: Effects of changing the tuning with TBspeakers driver unit A.

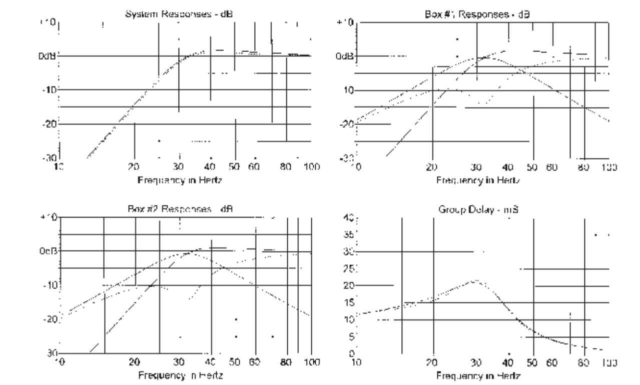

FIGURE 6: Other system performance for TBspeakers W8-704C (#197) driver pair.

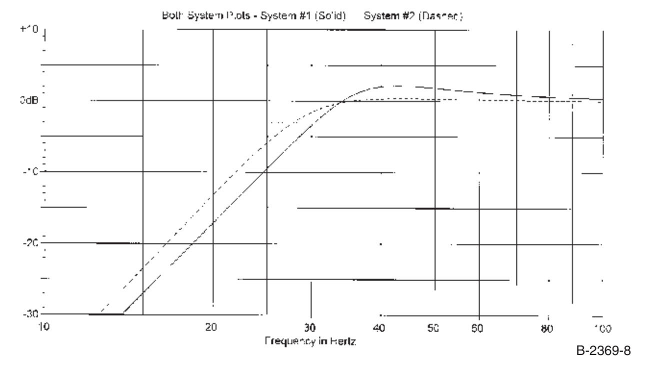

FIGURE 8: Effects of changing the tuning with TBspeakers driver unit B.

peaking from 100 down to 40Hz.

Figure 6 shows that the vent output dominates below 45Hz and the group delay looks reasonable. *Figures 7* and *8* show that tuning variations from 30 to 35Hz for both samples produce some shape change with f_3 ranging from about 28 to 30Hz depending on how you define it. The computed duct lengths for this design at f_B = 32Hz are 36″ long for a 4″ ID duct and 20″ for a 3″ ID duct.

We designed and built enclosures for both the MCM #55-2421 (#194) and TB speakers W8-704C (#197) drivers. Both systems yielded useful small subwoofers; however, we thought the more interesting system was that using the TBspeakers W8-704C (#197) driver. The remainder of this article covers that system.

PHOTO 6: Front view of TBspeakers W8-704C (#197) driver.

VERIFYING THE DESIGN

We tested TBspeakers driver unit A in a net test box volume of 0.818ft^3. *Figure 9* shows the input impedance plot indicating f_{MIN} at 31.5Hz. *Figure 10* shows the cone and port near-field

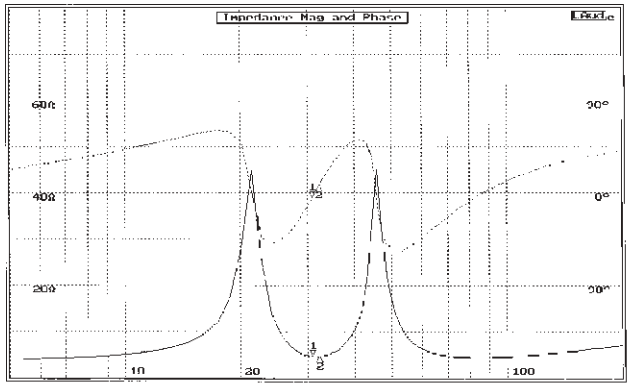

FIGURE 9: Input impedance for test of TBspeakers driver unit A in a VB.

magnitude responses set to the proper relative level. With f_B slightly below 30Hz (measured at 29.6Hz on screen), the box is a bit big and tuned a bit low. Note the port dominates the system output below about 43Hz as the design predicted (*Fig. 6*).

The system response (*Fig. 11*) is about as predicted up to 80Hz. The response droop above 70Hz is probably built right into the driver. This is a nice-looking response usable from 29Hz to about 100Hz within 3dB. You should keep the upper crossover limit near 100Hz with this driver.

The measured system group delay (*Fig. 12*) is noise free and thus valid above about 25Hz. This curve is worse than predicted, possibly because the test box is a bit big and tuned a bit low.

ENCLOSURE DESIGN

This design requires a net box volume of 0.8ft^3 with f_B near 32Hz. Using 4″ ID pipe for the vent gave an estimated length of 36″. We tried to implement such a port using 4″ ID PVC thin-wall (drain) pipe. Any box using this port diameter was going to end up bigger than our previous 12″ driver IB subwoofer!

We decided on the fallback approach using 3″ ID pipe with end flares. We obtained 3″ ID Precision Port™ kits, each of which consists of an external flare that has flange screw holes for mounting, an internal flare, two connecting collars, and a 12″ length of 3″ ID pipe. All pieces are listed as made of ABS plastic. *Photo 8* shows the pieces for a Precision Port™ kit and one kit assembled.

Clearly the straight port kit is not long enough as shipped; we would need to "fold-up" a longer duct. We found that thin-wall 3″ ID PVC drain-pipe elbows were somewhat compatible with the flares: The ABS and PVC pipe IDs are the same, but the ODs are different. With a bit of work you can mate the parts to yield a port duct with a smooth inside wall.

Precision Sound Products gives an equation for computing the needed duct length from end to end with the flares. For a net box volume of 0.8ft^3, you need an overall physical duct length of about 22″ for f_B = 32Hz and 25″ for f_B = 30Hz. We did not know at the time whether their equation factored in damping for the box.

Note that these lengths are a bit longer than those predicted by the Weems equations for a straight 3″ pipe in a damped box. Reference 5 had reported that using the flared ends required a duct slightly longer than a straight duct producing the same tuning.

We purchased numerous 3″ ID thin-wall (drain) PVC fittings and developed coiled ports of about 25″ total length. Two approaches looked reasonable.

One featured the port exit on the box rear with the duct coiled against the bottom of the box right up to the front panel and then part way back toward the rear. The driver would sit

PHOTO 7: Back view of TBspeakers W8-704C (#197) driver.

PHOTO 8: Precision Port™ components.

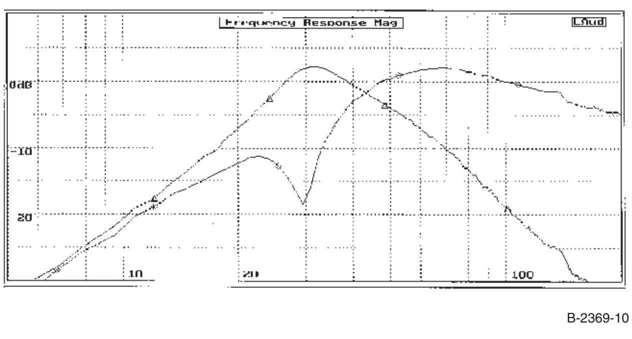

FIGURE 10: Near-field cone and port outputs for TBspeakers driver unit A VB test.

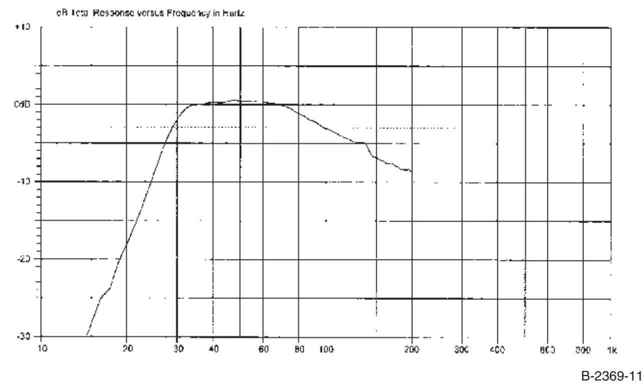

FIGURE 11: System response for TBspeakers driver unit A VB system.

above the duct. The box designed with this approach yielded a net box volume that was somewhat too big ($0.87ft^3$) and with overall dimensions of 13″ high, 13¼″ wide and 15-⅝″ deep. The net box volume is computed by subtracting the volume lost to the driver and vent duct assembly from the box gross volume.

The second design included the port exit at the side of the box with the duct coiled up behind the driver. The driver would then stand "clear" at the front of the box. With a net box volume of $0.8ft^3$ and a 25″ total duct length, the external box dimensions were 14″ high by 14¾″ wide by only 12⅜″ deep.

The W8-704C (#197) driver has a very large diameter magnet structure (*Photo 7*), and this box design gives more room for the rear of the cone to breathe. We considered this one to be the better design, so we built it. The duct design was such that it could easily be shortened from 25″ to 23″, but any further shortening would be a big problem.

BASIC CONSTRUCTION

The boxes are constructed with ¾″ thick industrial quality particleboard with appropriate stiffeners added. We believe that heavy, powerful drivers should be tied tightly to a very stiff front panel. We mounted the drivers with 10-24 machine screws into T-nuts. The front panel has stiffeners that tie it to the box corners and walls and that are positioned to retain the T-nuts which do not "bite" well into particleboard. This bracing may seem excessive, but our experience is that producing fast and taut bass requires keeping the driver from moving and the box from "ballooning."

We used yellow glue and 1⅝″ long panel nails to build the box. Where nails passed through a particleboard panel, we drilled a hole with a #51 bit (0.067″) to assure that the nail traveled straight into the edge of the mating panel. We set all nails in from outside panels below flush to permit sanding the panel.

The basic building approach was to cut out and fabricate all the major box panels. We then assembled the basic panels, except for the tops, and working through the open top installed most internal items, including assembly of the vent duct. To provide the absorptive damping included in the design, we covered the walls with a 1″ thickness of Owens-Corning (OC) #705 material. You can substitute your favorite VB damping approach.

Finally, we installed the top, cut oversized, and then routed it flush with the side walls. We then installed the stiffeners and damping material associated with the top through the driver hole.

The simple way to build a pair of subwoofer boxes is to purchase an industrial-quality particleboard sheet (which is actually 49″ by 97″) and have the lumberyard cut off two 97″ long strips 12½″ wide. You can cut the remainder into two reasonable strips for easy handling. Having the sheet cut into three 12½″ wide strips provides spare material if you make a cutting mistake. You can cut all major panels for the #197 subs from the 12½″ wide strips and fabricate the other pieces from the remaining particleboard.

FRONT PANEL CONSTRUCTION

To tie the driver securely to the front panel (FP), we mounted it with 1″ long 10–24 pan head machine screws and T-nuts. With 8″ drivers the clearances become tight. It is critical to position the mounting screw holes so that the screw heads clear the lip on the outer edge of the pressed-metal driver rim. This means pressing the bolt heads into the gasket on the driver's front during installation, but the gaskets seemed to recover just fine. The T-nuts also start to

PHOTO 9: Rear view of front panel with driver mounted.

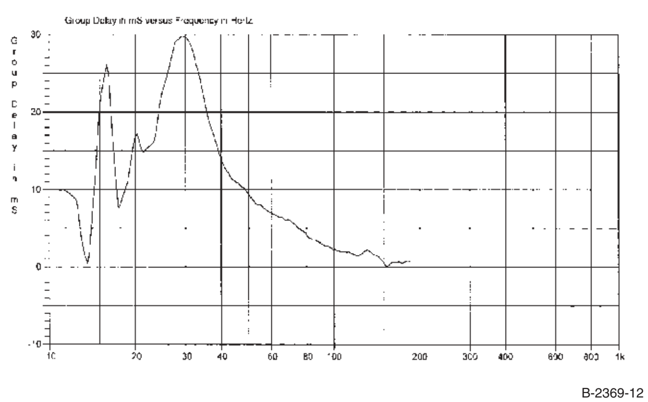

FIGURE 12: System group delay for TBspeakers driver unit A VB system.

intrude into the hole for the driver, but do not hit the driver so this presents no problem.

With high-displacement drivers it is important to relieve the back of the FP so the cone's rear can breathe. The W8-704C (#197) driver throws you a bit of a curve by having six frame ribs and eight mounting holes, so some ribs do not line up with mounting points. The solution was to relieve the driver hole everywhere between the T-nuts.

Our approach to constructing the FP was as follows:

1. Cut a driver-mounting hole of the proper diameter for the intended driver at the correct location on the FP. Note the boxes are built as mirror image pairs in terms of woofer offset from the vertical centerline.
2. Mark the location of the eight mounting screw holes. Orient the drivers so the mounting holes fall on horizontal and vertical lines. Be sure to make the holes on a small enough circle (about 7.7″ diameter for our hardware) so the machine screw heads will clear the outer lip of the pressed-metal driver rim. Our 10–24 pan head machine screws had a 0.365″ diameter head.
3. Drill a #10 (0.193″ diameter) hole at each mounting screw location. This will be snug on a #10 machine screw.
4. Verify that you can install the mounting screws through the driver and FP and secure them with the T-nuts on backwards. Outline the area on the back of the FP covered by each T-nut.
5. Remove the driver and relieve between the T-nut locations. We used a routing bit that cuts at a 45° angle to a depth of 0.430″ for both drivers. This depth takes you down to the start of the rear openings in the driver frame.
6. Drill the back of the mounting screw holes to the proper diameter and depth to take the T-nuts. Our T-nuts required a 0.250″ hole.
7. Install the T-nuts so all three spikes grab particleboard. We pressed the T-nuts in with a vise after installing a ¾″ long machine screw to assure they stayed aligned with the #10 hole.
8. Verify that you can mount the driver. It is much easier to fix any problem now than when the FP is mounted.

The FP/box stiffeners are designed so that they will retain the T-nuts, but this is not practical on the two T-nuts oriented at 45º on the wide side of the FP. To retain these T-nuts, we installed #6 by ⅜″ pan head sheet metal screws next to the T-nut and tightened the screws just enough to retain them.

Photo 9 shows a view of the back of the FP with the driver mounted showing the T-nuts and relieving and demonstrating the offset between frame ribs and mounting points.

Next issue we conclude construction of the subwoofer boxes. ❖

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6. David Weems, personal communication.

SOURCES

Creative Sound Solutions, 32-32691 Garibaldi, Abbotsford, BC Canada, 604-504-3954, www.creativesound.ca

MCM Electronics, 650 Congress Park Drive, Centerville OH 45459, www.mcmeletronics.com, 937-434-0031

Nuera Acoustic Technology, 6538 Albery Place, Burnaby, BC Canada, V5E-4G2, 604-517-6683, www.nuera-acoustic.ca

Representative in the continental US: 604-805-3867, between 9am and 7pm PST, FAX 604-517-6693 (our source for TBspeakers drivers)

Owens-Corning - For information on obtaining the #705 damping material call the Customer Service Center: 800-328-7617

Parts Express, 725 Pleasant Valley Dr., Springboro, OH 45066-1158, 1-800-338-0531, www.partsexpress.com

Precision Sound Products, 815-599-0662, www.psp-inc.co

RAW Acoustics, 16756-85 Ave., Surrey, BC Canada, V4N-4W3, 604-576-8951, www.rawacoustics.ca

Ultimate Sound, 437 W. Grimes Creek Rd., Bayfield, CO 81122, 1-888-900-9687, www.ultimate-sounds.com (Our source for MA-Audio drivers)

A Bottom Octave Compensator

This simple equalizer can compensate a typical sealed-box woofer to be anechoically flat to any arbitrary low frequency—20Hz, in the example shown. **By Dennis Colin**

I'm angry and I'm not going to take it! I'm tired of "subwoofers" that are –3dB at 40Hz, or even higher! This is what normal woofers used to do since about 1958, courtesy of the Acoustic Research AR-3 and fellow clones. Now, with a plethora of mini-monitors, albeit of excellent midrange and treble quality, you have "woofers" with an f_{-3} of 70Hz or so (of course, the biggest joke is "computer subwoofers" that, if lucky, can squeak out a little bit of 60Hz!).

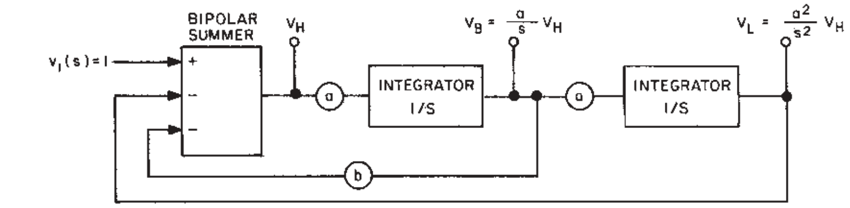
When not listening at the maximum volume your system is capable of, you can equalize most decent sealed woofers to compensate their 12dB/octave (anechoic) rolloff to be flat to 20Hz. Note: Room gain doesn't take effect in typical rooms until below 30Hz. And if the speaker is anechoically flat to 20Hz, a little bit of boost at 20Hz (5dB in my room) is welcome; this helps your hearing rolloff at less than concert-hall volume (Fletcher-Munson, et al.).

There is actually more lowest-octave content in recorded material than is often believed. Besides pipe organs and synthesizers, for example, bass drums and tympani can have significant transient-excited sideband content at and even below 20Hz. You can feel (if not quite hear) this in a concert hall or large club venue; it gives a sense of very deep ambience, like rolling thunder. So why not reproduce it?

Warning! This compensator generates a 12dB/octave rise below the sealed woofer's box frequency (fb), equalizing the woofer's 12dB/octave (2nd order) rolloff slope down to a selectable frequency. Do not use this with vented systems! Besides having twice the rolloff slope, the woofer cone is unrestricted (except by the suspension) below the system rolloff frequency. Applying high power with boosted ultra-low frequencies could easily damage a vented woofer.

CIRCUIT HISTORY

Figure 1 is just one variant of a circuit old enough to be called historic. In 1966, my EE professor called it a "2nd order differential equation solver," and that it is (they had vacuum tube op amps then, made by George Philbrick!). Around 1971, while designing the ARP2600 music synthesizer, I presented a paper to the Audio Engineering Society with a title suggested by David Friend of ARP: "The Electrical Design and Musical Applications of a Combination Voltage Controlled Highpass, Bandpass, Lowpass, Band Reject Filter Resonator." (I would have entitled it "A Voltage Controlled Filter," but Dave was the marketing manager!) This circuit used the "new" solid-state op amps, along with analog multipliers for voltage control of



$S = j\omega = 2\pi jf$

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$$V_H = 1 - bV_B - V_L = 1 - \frac{ab}{s} V_H - \frac{a^2}{s^2} V_H$$

$$V_H \left(1 + \frac{ab}{s} + \frac{a^2}{s^2}\right) = 1, V_H = \frac{1}{1 + \frac{ab}{s} + \frac{a^2}{s^2}}$$

$$V_H = \frac{s^2}{s^2 + abs + a^2} = \text{ Highpass, gain} = 1 \text{ @ } s = \infty$$

$$V_B = \frac{as}{s^2 + abs + a^2} = \text{ Bandpass, gain} = \frac{1}{b} \text{ @ } s = \pm ja$$

$$V_L = \frac{a^2}{s^2 + abs + a^2} = \text{ Lowpass, gain} = 1 \text{ @ } s = 0 \text{ (0C)}$$

Denominator poles: $\omega_o = a, \ f_o = \frac{a}{2\pi}, Q = \frac{1}{b}$

At $f = f_o$, gain of all 3 outputs = Q; dB gain = 20log Q

FIGURE 1: State-variable filter (Bi-quad).

rolloff frequency and Q. ARP made it their module 1047; one was bought by The Who and used in their song about "We Won't Get Fooled Again."

PHOTO 1: The bottom octave compensator.

PHOTO 2: Potted all-discrete multipliers.

Photo 1 shows the unit; *Photo 2* shows the potted all-discrete multipliers. *Photo 3* shows a similar filter I just breadboarded with today's components, using the CA3280 operational transconductance amplifiers to achieve voltage-controlled filter frequency range of 1Hz to 74kHz. *Photo 4* shows this filter's notch output used to display the 12.027% THD of a symmetrical triangle waveform.

Today this circuit is often simply called a "Bi-quad." That's because its Laplace transfer function is the ratio of two quadratic polynomials in $s = j\omega + \sigma$ ($\omega = 2\pi * \text{frequency}$; σ is the damping coefficient). The circuit allows adjustment of all six polynomial coefficients. With variable coefficients, the circuit is sometimes referred to as a "state-variable filter."

STATE-VARIABLE FILTER—BASIC TOPOLOGY

Figure 1 shows a block diagram configured for independent control of f_o (rolloff or center frequency) and Q. The input ($V_1(s)$ in Laplace terminology) is set to 1, which means a flat spectrum unit impulse stimulus. This is fed through a bipolar summer (no, not a psychotic vacation) and a gain coefficient "a" to an integrator (such as an inverting op amp with resistive input and capacitive feedback). Negative feedback is applied back to the input through coefficient "b." This integrator output also feeds a second coefficient "a" and a second integrator, the latter also negatively fed back to the input, but with unity gain coefficient.

The multiple integrator feedback loops produce highpass (V_H), bandpass (V_B), and lowpass (V_L) outputs, all with a common quadratic (2nd order) denominator. *Figures 2a, b,* and *c* show decibel versus log frequency (Bode) plots. Note that coefficients "a" determine f_o, while "b" determines Q. Making "a" and "b" voltage-controllable, as with analog multipliers, allows independent and remote control of f_o and Q.

OUTPUT SUMMING

Figure 3 shows that by summing the outputs with adjustable coefficients (H, B, L), the transfer function V_0/V_1 (s) is the ratio of two quadratic polynomials (biquad). Some of the many available responses are:

1. Highpass, bandpass, or lowpass; when two of the three coefficients "H," "B," "L" are zero.
2. Infinite notch, when "B" = 0.
3. Unity-Gain Allpass (phase shift with flat amplitude), when H = L = 1 and B = −1.
4. Shelf responses (*Figs. 4a, b*) when H ≠ L. Various combinations of peaking (at f_p) and dipping (at f_z) are possible, adjustable by Q_p and Q_z, respectively.

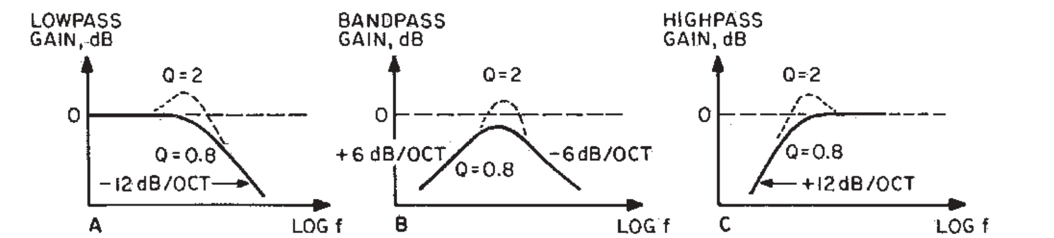


FIGURE 2: Bode plot responses. A-2262-2

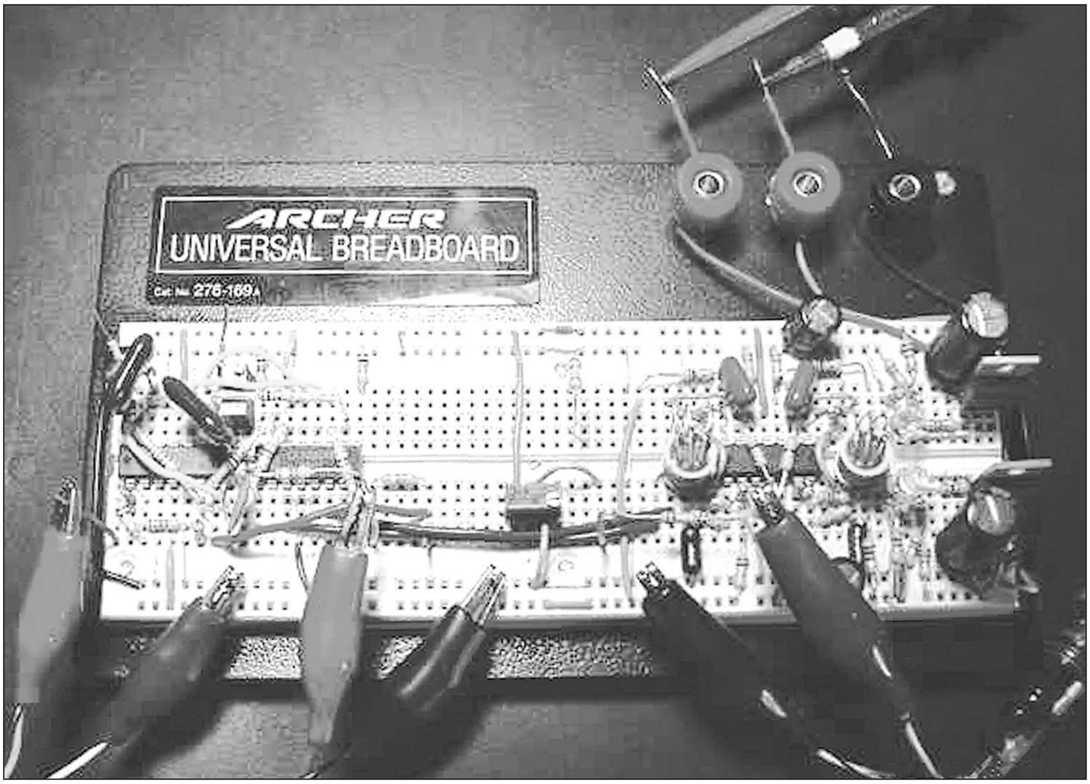
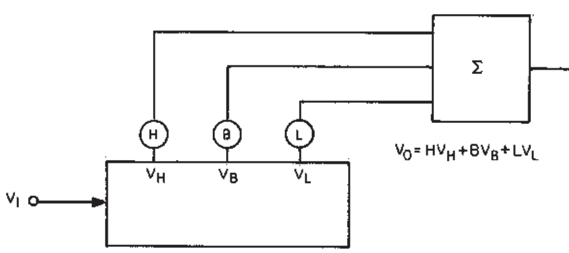


PHOTO 3: A similar filter breadboarded with today's components.



$$\frac{V_o}{V_1}(s) = \frac{HS^2 + Bas + La^2}{S^2 + abs + a^2}$$

Numerator (zeroes): $f_z = \frac{a}{2\pi}\sqrt{L/H}, Q_z = \frac{\sqrt{HL}}{B}$

Denominator (poles): $f_p = \frac{a}{2\pi}, Q_p = \frac{1}{b}$

DC Gain (f = 0) = L

HF Gain (f = ∞) = H

If B = 0, but not H and L, response is infinite notch (zero gain) @ f_z

FIGURE 3: State-variable filter output summing. A-2262-3

The response type in *Fig. 4a* is used for the sealed-box woofer bass extender (bottom octave compensator) to be described.

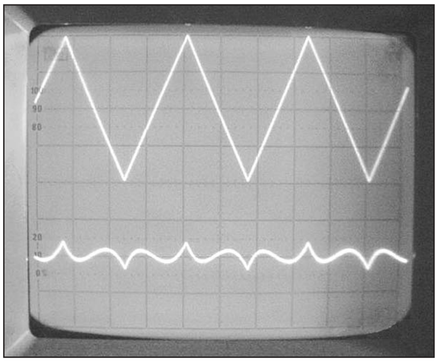


PHOTO 4: Notch output from *Photo 3*.

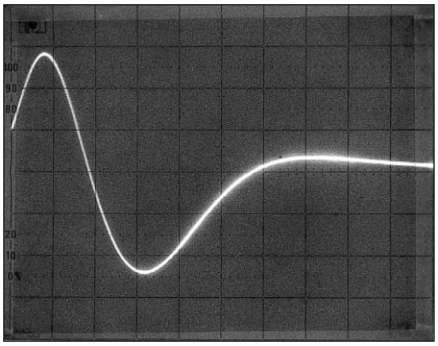


PHOTO 5A: Compensator of *Fig. 5*.

ADDITIONAL HIGHPASS (HP) ROLLOFFS

In the circuit of *Fig. 5*, C1 and C6 provide a net 2nd order HP rolloff, which, combined with the *Fig. 4a* response and the woofer rolloff, results in an overall 4th order anechoic LF rolloff response at a subterranean frequency of your

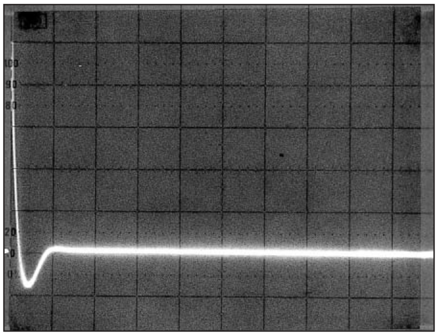


PHOTO 5B: Sealed woofer (simulated) f_b = 40.5Hz, f_{-3} = 38Hz, Q_b = 0.75.

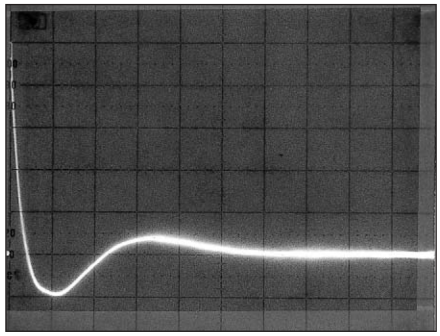


PHOTO 5C: Compensated woofer.

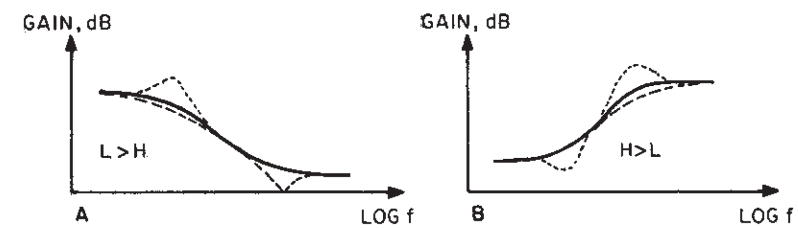


FIGURE 4: Shelf responses.

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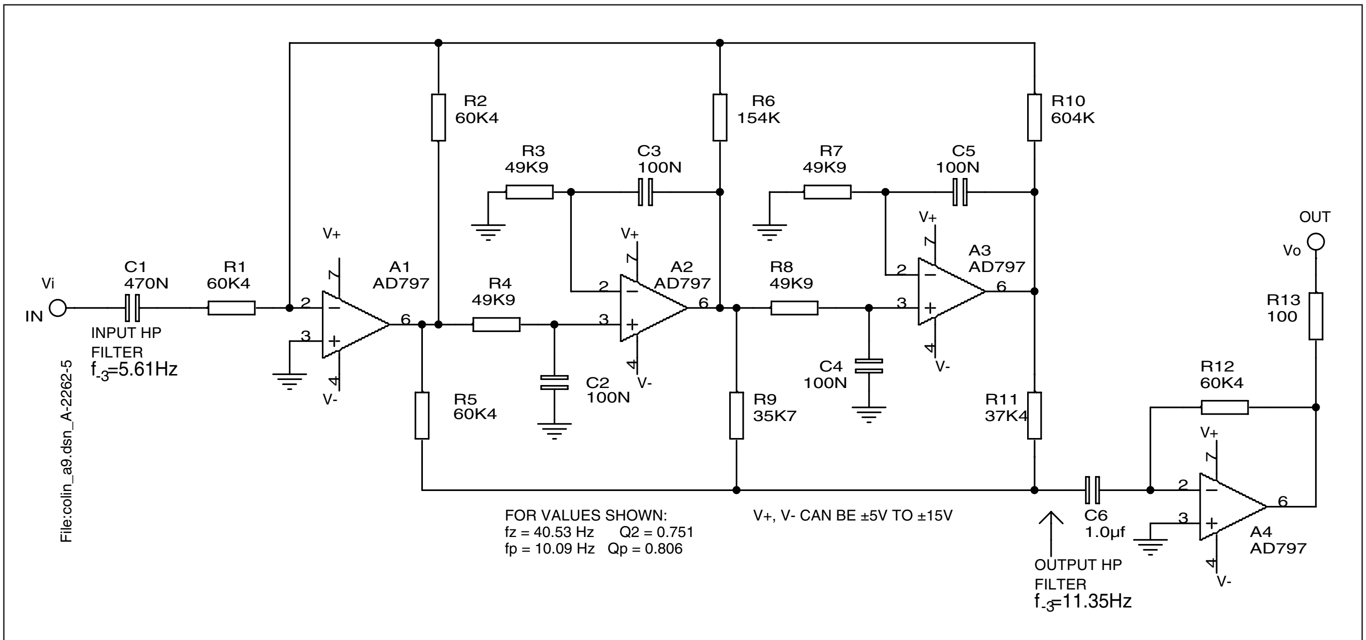

$$-H = 1 + B\frac{R_H}{R_B} + L\frac{R_H}{R_L}, -H = 1 + \frac{HR_H}{R_B RCS} + \frac{HR_H}{R_L(RCS)^2}$$

$$H\left[1 + \frac{R_H}{R_B RCS} + \frac{R_H}{R_L(RCS)^2}\right] = -1, H = \frac{-(RCS)^2}{(RCS)^2 + \frac{R_H}{R_B}RCS + \frac{R_H}{R_L}}$$

$$V_O = \frac{(RCS)^2 + \frac{R_H}{R_{BO}}RCS + \frac{R_H}{R_{LO}}}{(RCS)^2 + \frac{R_H}{R_B}RCS + \frac{R_H}{R_L}}$$

Denominator (poles): $\omega_p = \frac{\sqrt{R_H/R_L}}{RC}, f_p = \frac{\sqrt{R_H/R_L}}{2\pi RC}$

$$Q_p = \frac{R_B}{\sqrt{R_H/R_L}}$$

Numerator (zeroes): $\omega_z = \frac{\sqrt{R_H/R_{Lo}}}{RC}, f_z = \frac{\sqrt{R_H/R_{Lo}}}{2\pi RC}$

$$Q_z = \frac{R_{Bo}}{\sqrt{R_H R_{Lo}}}$$

$$Q_z = \frac{R_{Bo}}{\sqrt{R_H R_{Lo}}}$$

@ $F = \infty, V_o = 1$

If $A(s) = as^2 + bs + c$, $\omega_0 = \sqrt{c/a}$ $Q = \frac{\sqrt{ac}}{b}$

In *Fig. 5*, $R_H = R1 = R2 = R5 = R12$
$R_B = R6$, $R_L = R10$, $R_{BO} = R9$, $R_{LO} = R11$
$R = R3 = R4 = R7 = R8$, $C = C2 = C3 = C4 = C5$

FIGURE 5: Compensator schematic.

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choosing. Values in *Fig. 5* and the following design procedure produce a well-damped rolloff to avoid excessive infrasonic ringing. *Figure 6* shows responses for the values in *Fig. 5* with a woofer f_b = 40.5Hz, Q = 0.750.

PRACTICAL CIRCUIT DESCRIPTION

Figure 5 shows a circuit designed specifically for sealed woofer compensation. It is an active equalizer with unity gain in the flat region above the compensated LF response, to be placed just before the power amp. Of course, you should use high-quality op amps and capacitors.

A2 and A3 are non-inverting integrators. While these require two caps each, they avoid the need for additional inverting amps to restore overall positive polarity. R2, 6, and 10 provide the feedback of direct, integrated, and doubly-integrated signals as shown in *Fig. 1*.

R5, 9, and 11 perform the signal summing shown in *Fig. 3*; these resistors are inversely proportional to the coefficients H, B, L (*Fig. 3*), respectively. A4 is an output signal-summing amp. C1 and C6 provide the additional high-pass (HP) rolloffs previously described. *Figure 7* is an analysis of this circuit.

DESIGN PROCEDURE

1. Choose R1 = R2 = R5 = R12, R3 = R4 = R7 = R8, C2 = C4 = C5 (suggested values in *Fig. 5*).
2. Select f_p = LF rolloff of compensated speaker, Hz ≈ compensator peak gain frequency.
3. Set $R_{10} = R_1/(2\pi R_3 C_2 f_p)^2$ Note: Component designator numbers in subscripts to avoid math confusion.
4. Choose Q_p; Q_p = 0.800 for response shown.
5. Set $R_6 = Q_p \sqrt{(R_1 R_{10})}$
6. Select f_z = start-of-boost frequency, Hz. This is not f_{-3} of speaker, unless its $Q = 1/\sqrt{2} = 0.707$. It is the intersection of the speaker's asymptotic 12dB/octave rolloff slope (straight line on Bode plot) with 0dB speaker reference in its flat region, also called f_b.
7. Set $R_{11} = R_1/(2\pi R_3 C_2 f_z)^2$.
8. Determine Q_z = sealed woofer Q.

9. Set $R_9 = Q_z \sqrt{(R_1 R_{11})}$.
10. Set input RC rolloff: $f(in)_{-3} = 1/2\pi R_1 C_1$.
    Set $f(in)_{-3} = 0.556f_p$; $C1 = 0.286/R_1 f_p$ (farads, ohms, Hz).
11. Set output RC rolloff: $f(out)_{-3} = 1/2\pi R_0 C_6$, where $R_0 = R_5$, R_9, and R_{11} in parallel; set $f(out)_{-3} = 1.125f_p$; $C_6 = 0.141/R_0 f_p$ (farads, ohms, Hz).
12. Build it and enjoy bottomless natural bass!

TRANSIENT RESPONSE TESTS

I looked at square-wave responses using an actual sealed woofer with f_b (not f_{-3}) of 40.5Hz and Q of 0.75, the same parameters as the circuit of *Fig. 5*. Responses were much like a passive LCR circuit I also used (to simulate the woofer), but the latter was of course free of room effects (I'm using spectral components below 10Hz!). So I used the LCR circuit for the photos shown.

Photos 5a, b, and *c* are step responses (a 2.5Hz square wave was used; responses to positive step shown). Time scales are 20mS/division (200mS total span). *Photo 1a* is of the compensator alone, *Photo 1b* is the sealed woofer simulator, and *Photo 1c* is the combination (compensated woofer).

Note that the compensator (*Photo 5a*) response starts with a positive one division jump (representing the unity-gain HF response), followed by a "swept-frequency" cycle; the latter spectrally "contains" the boosted region in *Fig. 6* of about 2.5–40Hz.

The negative-going overshoot in *Photo 5b* is the minimum-phase consequence of the (simulated) woofer's 2nd order rolloff below 40.5Hz. Note how in *Photo 5c* the compensator simply and smoothly extends the woofer step response's time scale. This follows from the woofer's f_{-3} being extended from 38Hz to 14Hz (*Fig. 6*).

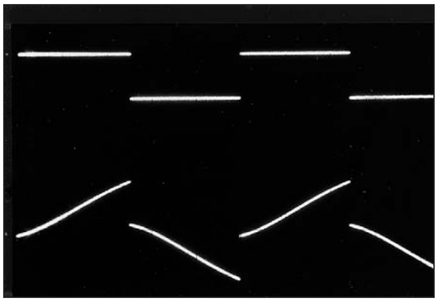

PHOTO 6A: Compensator square-wave response. Upper trace = input, lower trace = output. 40Hz.

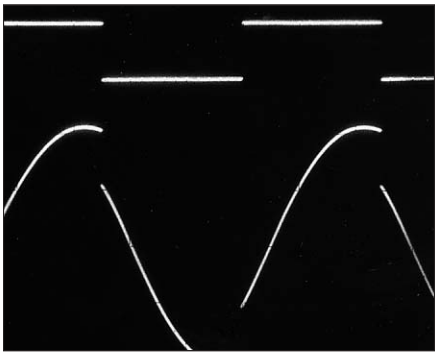

PHOTO 6B: 20Hz.

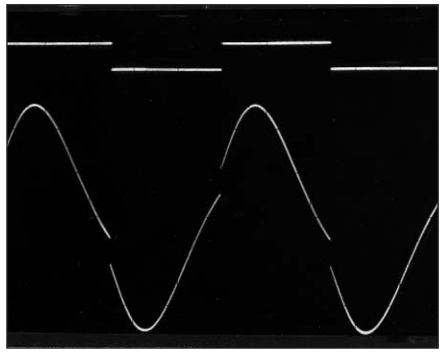

PHOTO 6C: 10Hz.

Photos 6a, b, and *c* show the input and output responses of the compensator to a square wave at 40, 20, and 10Hz, respectively. The gains of 11dB at 20Hz and 18dB at 10Hz are evident.

But the proof of good transient response is in the combinational pudding. *Photos 7a, b,* and *c* show the woofer responses uncompensated (upper traces) and compensated (lower traces), to a square wave at 40, 20, and 10Hz, respectively. The extension of low-frequency content is clearly evident.

Notice also that no ringing has been added. In fact, the 20 and 40Hz compensated square-wave response shapes have been improved to a nearly ideal exponential decay. And while the overall frequency response is nearly flat (–1dB) to 20Hz, it's the (minimum) phase shift of the components ≥ 20Hz (due to the < 20Hz rolloff) that causes the exponential decay, even though the 20Hz square-wave input has no spectral content below 20Hz. A Fourier analysis of the lower *Photo 7b* waveform would show the square wave's odd harmonic series (20, 60, 100, 140 . . . Hz) with the 20Hz fundamental amplitude 1dB lower than in a perfect square wave, relative to the harmonics.

PHOTO 7A: Woofer square-wave responses. Upper trace = uncompensated, lower trace = compensated. 40Hz.

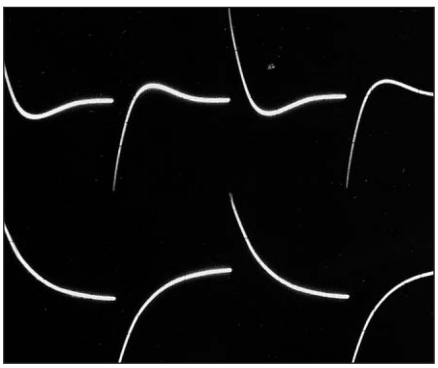

PHOTO 7B: 20Hz.

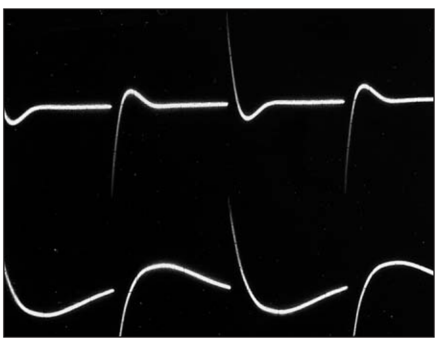

PHOTO 7C: 10Hz.

COMPENSATOR USAGE

For several years, I've used this type of compensator to extend response to 20Hz, –1dB, with two sealed woofer systems: (a) an 8″ Focal 8K416 (j) in a 0.91ft^3 box; f_b = 61Hz, Q_b = 0.54, and (b) a 10″ Scan-Speak 25W/8565-01 in a 1.6ft^3 box; f_b = 40.5Hz, Q_b = 0.75, f_{-3} = 38Hz. Available continuous power is 50W for the 8″ Focal, 100W for the 10″ Scan-Speak. Some observations:

1. For nearly all music except the deepest pipe organs and exaggerated synthesizer bass, I can drive the amp to clipping; what clips is not the boost-compensated ultra-low bottom octave. Most music doesn't have enough 20–40Hz content for this to clip the amp, even with 18dB boost at 20Hz.

   However, much music has sufficient 20–40Hz content to hear and

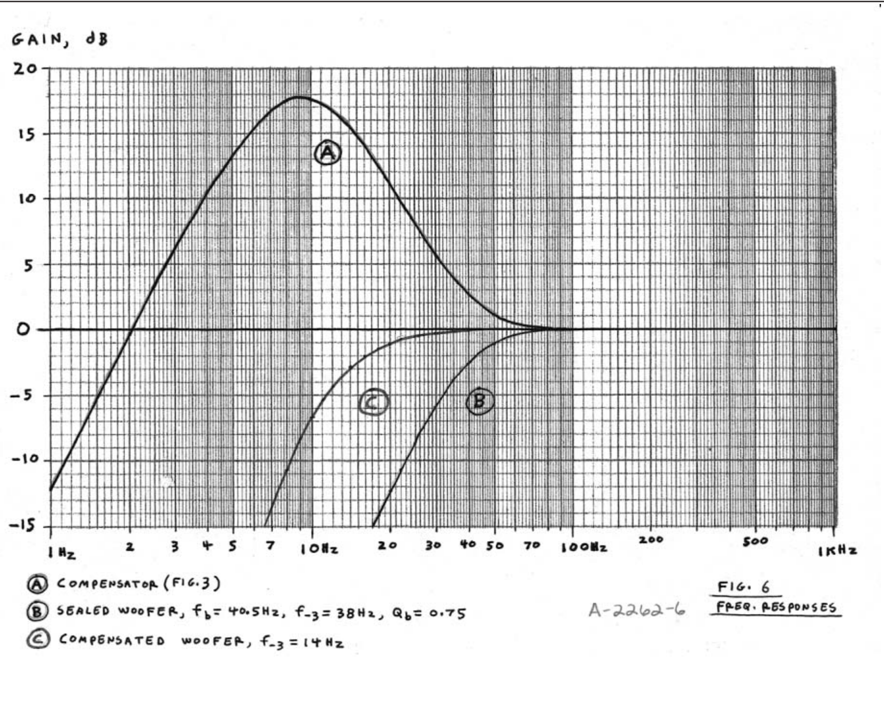

FIGURE 6: Frequency responses.

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IMPORTANT SPECIFICATIONS

Woofer: Scan-Speak 25W/8565-01. Manufacturer data: f_s = 19Hz, V_{as} = 229 ltr, Q_{TS} = 0.845

Cabinet: 26″ × 12″ × 12″ outside, ¾″ wood

Internal vol. = 1.563ft^3 = 44.26 ltr (not counting driver and brace vol.; negligible)

Center brace panel, 10.5in^2 × ¾″, 7″ round hole.

Damping: three sides, bottom covered with Deflex rubber panels. Loosely filled with Acousta-stuff.

Calculated box parameters (not taking into account cabinet damping and adiabatic-to-isothermal resonance lowering): f_b = 47.2Hz Q_b = 0.845

Measured box parameters: f_b = 40.5Hz Q_b = 0.75 f_{-3dB} = 38Hz.

Op Amps: I used TL074 Quad op-amp. For lowest noise and distortion, use Analog Devices AD797—expensive: $9.48 ea. (Newark)—but very good; Noise = 0.9nV/√Hz (open-circuit thermal noise of 50Ω resistor!)

THD = –120dB (0.0001%) at 20kHz

Note: Only A1 and A4 pass full-range audio, so you could use AD797 for those, and (dual) TL072 for A2, 3.

Supply voltage: Recommend ±15V, but will work down to ±6V.

feel when accurately compensated for speaker rolloff. Examples include bass drums, tympani, and even some acoustic string bass when close-miked; the latter can produce transient-excited spectral content well below the lowest fundamental (open E string at 41.2Hz). Likewise, but more so, for an electric bass plugged directly into a recording console.

2. Of course, when playing recordings of pipe organs, synthesizers, or any source of sustained ultra-low bass, the amp will certainly clip on these boosted components. But at a reasonably lower-than-full volume, these sources benefit greatly by having flat response to 20Hz. In fact, with satisfactory reproduction of these foundation frequency fundamentals, you don't feel such a need to crank up the juice to approach live music levels.
3. After years of pounding these (high quality) woofers, a pair of each, none has blown nor shown signs of abuse.
4. With anechoic flatness to 20Hz, room gain produces a net acoustic boost at 20Hz. For example, with the 10″ Scan-Speak systems in my 3000 ft^3 living room, 20Hz room gain is about 5dB. I've never found this objectionable, because at lower-than-live levels the ear needs considerable Fletcher-Munson (loudness) compensation at 20Hz. Furthermore, I find that the type of compensation I describe here has none of the boominess, muddiness, nor other colorations rampant in typical "loudness compensation" circuits.
5. I use the 8″ Focal systems in my car (see "Real Backseat Bass," *Speaker Builder* 7/98). Here, the cabin (room) gain starts at around the 61Hz f_b of the woofer, so acoustic response is flat (to ≈ 12Hz) without electronic compensation. However, I usually use a compensator that flattens the woofers anechoically to 20Hz. The result is an in-car response boost of about 20dB at 20Hz. Even this doesn't sound boomy or muddy. Perhaps in addition to loudness compensation, such boost is needed to keep up with wind and road noise, which can be very strong at 20Hz.

CONCLUSION

I have described a simple equalizer circuit that can accurately extend the LF response of any high-quality sealed woofer with straightforward design rules. You can arbitrarily select the compensated (extended) LF rolloff. For high-quality 8″ and larger sealed woofers, flat response to 20Hz is practical. But smaller woofers can also be extended within reason (a 4″ woofer can be extended perhaps to 40Hz, but not to 20Hz, for example).

One caution: with LP playback and anything less than excellent vibration isolation, or with warped LPs, infrasonic rumble would be greatly amplified with flat-to-20Hz compensation, so you should not use this. But with clean playback sources, and especially with 10″ and larger sealed woofers, having flat bass response to 20Hz will add a very satisfying and natural sense of bottomless depth and solid foundation to many recordings. ❖

Build a Universal PC Sound Amplifier

This PC amp delivers 2 × 20W power and insulated input for distortion-free sound; volume and tone controls plus standby switch on the front panel; and earphone, PA driver, recorder output, and aux input.

By Jeno Keceli

You can only marvel at the advertised gigantic power of PC systems. It is known in professional circles that these figures do not really mirror actual power. Lack of control and connections of these products strongly limit their application. A good solution is to connect an external amplifier to the output of the soundcard. However, this would take too much room on the already cramped computer desk. The number of the various wires would increase and the interference could not always be filtered out.

If you consider these limitations, it becomes clear that it is advantageous to have a high-quality sound amplifier inside the PC box. This article shows how you can build such an amplifier. I will analyze the possible pitfalls and how to overcome them and also refer to the possible changes that can enable the universal implementation of the amplifier.

PROBLEMS AND SOLUTIONS

You must first consider the output power, which largely depends on the supply voltage. With only 12V at your disposal from the PC power supply, this, similar to that of the car radios, reduces the output power to approximately 4W. If you require higher power output than this, you need to either increase the supply voltage or apply bridge circuits with two pairs of amplifiers. Increasing the 12V is only possible with a separate DC-DC converter, which requires room and more components. Furthermore, there is a higher possibility of failure and, in case of non-adequate filtering, it is a source of interference.

Bridging the two pairs of amplifiers is more advisable, and is the solution I have chosen. This way you can also achieve the 2×20W music power. Anyway, it is not recommended to drive a higher load from the PC power supply, but it is advisable to choose a higher ca-

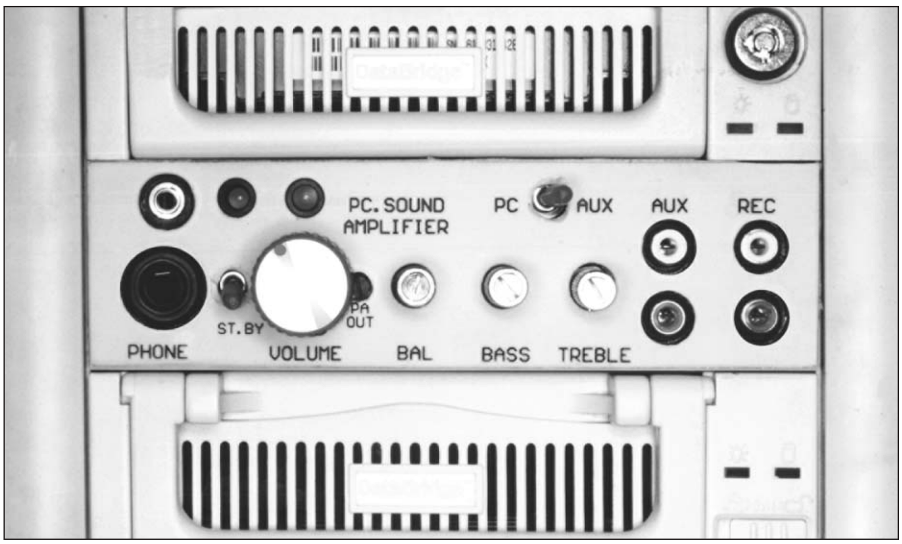

PHOTO 1: Front panel of PC-amp.

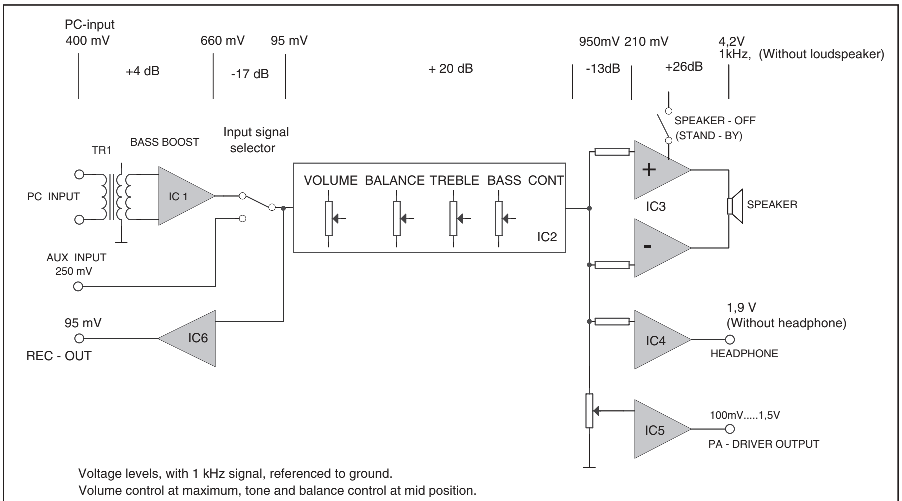

FIGURE 1: Block diagram.

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ABOUT THE AUTHOR

Jeno Keceli (of Hungary) is a long-time electronics enthusiast with 40 years of experience. His main fields of interest are designing and building tube and transistor power-amplifiers. He has much experience with sound and disco equipment and has published articles in electro technical journals. He now runs his own firm, where he builds, repairs, and gives advice. Email: keczeli@yahoo.de

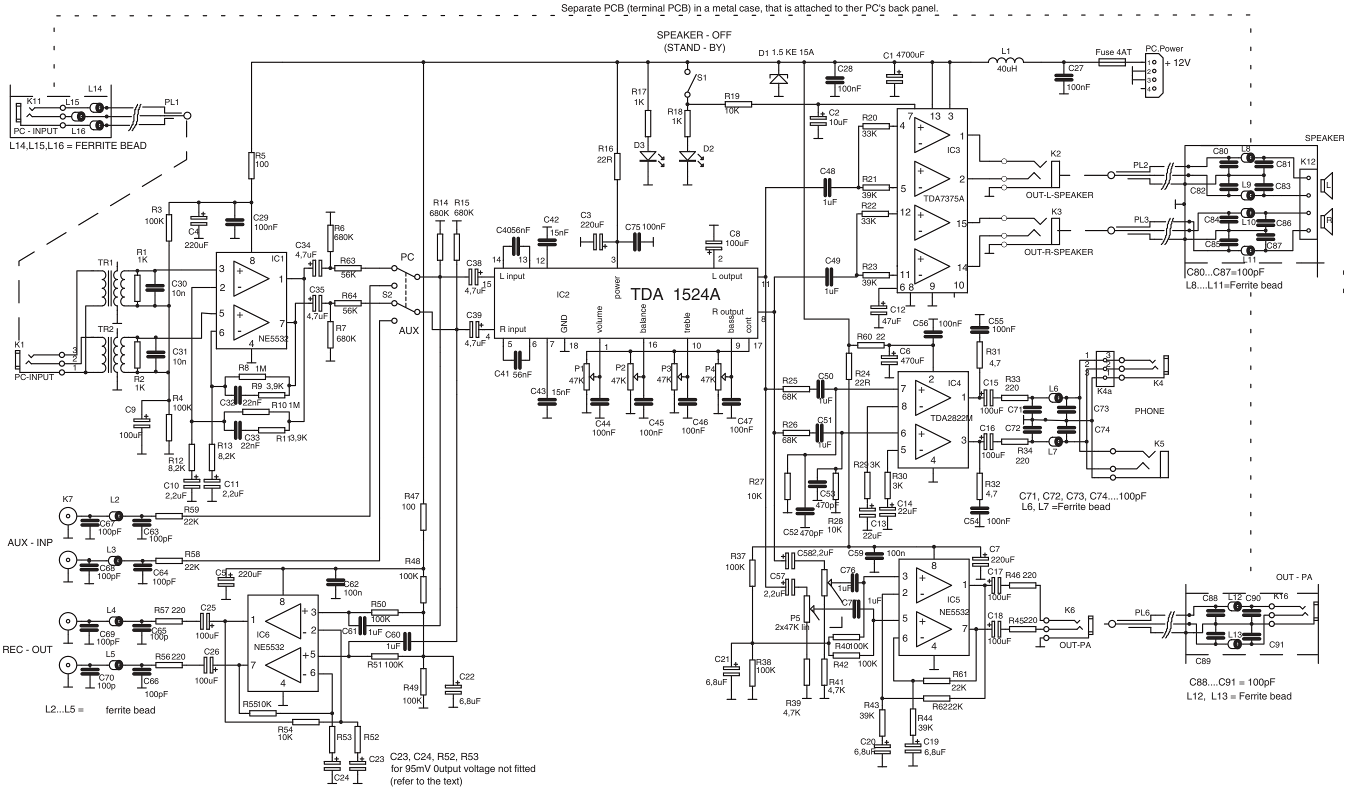

FIGURE 2: Amplifier's detailed schematic (that is attached to the PC's back panel).

pacity power supply. Of course, this all depends on the current internal load. In my case, the 12V supply, loaded with 12A, works perfectly. You also must take into consideration the current-consumption of the PC.

In the case of the bridging configuration, two amplifiers (per channel) are applied so that the loudspeaker is not connected to the ground, but to another amplifier that is controlled by a signal of the phase inverter. Hence, from the same supply voltage, you will obtain double output voltage swing and fourfold output power. For this application, you can obtain excellent ICs with two or four internal power amplifiers.

You can guess the second problem, which manifests itself only during the first test. From the loudspeaker, as in the case of a shortwave radio, the mixture of various peeping sounds disturbs pure musical pleasure. This is caused

TABLE 1
TEST RESULTS

Supply voltage:	+12V from PC		
Current drain	approximately 120mA (standby), approximately 3.5A (full power)		
Output power	2×20W at 4Ω, musical	2×15W	(THD 0.5%)
Frequency response	20Hz.....20kHz,	+1...–3dB	
PC-input, insulated	400mV	input impedance Approximately 600Ω	
Aux input	250mV	input impedance 47kΩ	
REC output	95mV	from preamplifier	
PA-driver output	100mV.....1.5V (approximately –15....+10dB) (adjustable with a screwdriver)		
Power output shorting and overheat protection control			
Headphones output cca. 100mW			
Standby switch, volume (built-in contour), tone and balance control on the front panel.			
Size: front panel 148×42mm (similar to a CD-ROM), depth 175mm			

PC – AMPLIFIER'S PART LIST

SCHEMATIC REFERENCE DESIGNATOR	PART TYPE VALUE	DESCRIPTION
IC1, IC5, IC6	NE5532	
IC2	TDA 1524A	
IC3	TDA7375A	
IC4	TDA2822M	
D1	1.5 KE 15A	Protection-Suppression Diode
D2	LED red, 5mm	
D3	LED green, 5mm	
Fuseholder	20×5mm/1″×¼″	PCB fuseholder
Fuse	4AT	Fuse, glass/ceramic
TR1, TR2	Audio transformer	600Ω/600Ω (refer to text)
C1	4700µF/16V	Electrolytic-Radial
C2	10µF/16V	Electrolytic-Radial
C3, C4, C5	220µF/16V	Electrolytic-Radial
C6	470µF/16V	Electrolytic-Radial
C7	220µF/16V	Electrolytic-Radial
C8, C9	100µF/16V	Electrolytic-Radial
C10, C11	2.2µF/16V	Tantalum
C12	47µF/16V	Electrolytic-Radial
C13, C14	22µF/16V	Electrolytic-Radial
C15, C16, C17, C18	100µF/16V	Electrolytic-Radial
C19, 20, 21, 22	6.8µF/16V	Tantalum
C23, C24	not fitted	refer to the text
C25, C26	100µF/16V	Electrolytic-Radial
C27, C28, C29	100nF	Ceramic
C30, C31	10n	Ceramic
C32, C33	22nF	Ceramic
C34, C35, C38, C39	4.7µF/16V	Tantalum
C40, C41	56nF	Ceramic
C42, C43	15nF	Ceramic
C44, C45, C46, C47	100nF	Ceramic
C48, C49, C50, C51	1µF	Ceramic/MKS/MKP
C52, C53	470pF	Ceramic
C54, C55, C56	100nF	Ceramic
C57, C58	2.2µF/16V	Tantalum
C59	100n	Ceramic
C60, C61	1µF	Ceramic/MKS/MKP
C62	100n	Ceramic
C63, C64, C65, C66		
C67, C68, C69, C70	100pF	Ceramic SMD or Disc
C71, C72, C73, C74	100pF	Ceramic SMD or Disc
C75	100nF	Ceramic
C76, C77	1µF	Ceramic/MKS/MKP
C80-91	100pF	Ceramic SMD or Disc
K1, K6, K11, K16	Phone-jack, 3.5mm stereo	Socket PCB mounted
K2, K3, K5	Phone-jack ¼″ stereo	PCB mounted socket
K4	Phone-jack, 3.5mm stereo	panel mounted
K4a	Terminal block	PCB mounted 3 pole
K7	4× RCA-jack PCB-mounted socket	
K12	PCB mounted stereo loudspeaker connector	

SCHEMATIC REFERENCE DESIGNATOR	PART TYPE VALUE	DESCRIPTION
PC-power	Four-way PC power connector, PCB-mount, male	
PL1, PL6	Phone plug 3.5mm stereo	
PL2, PL3	Phone plug stereo ¼″	
L1	40µH 4A	Inductor Axial
L2-L16	Ferrite bead 5mm	
P1, P2, P3, P4	47k lin, PCB-mounted	Potentiometer
P5	2×47k lin, PCB-mounted	Potentiometer
R1, R2	1k	All resistors are ¼W
R3, R4	100k	
R5	100Ω	
R6, R7	680k	
R8, R10	1M	
R9, R11	3.9k	
R12, R13	8.2k	
R14, R15	680k	
R16	22Ω	
R17, R18	1k	
R19	10k	
R20, R22	33k	
R21, R23	39k	
R24	22Ω	
R25, R26	68k	
R27, R28	10k	
R29, R30	3k	
R31, R32	4.7Ω	
R33, R34	220Ω	
R37, R38	100k	
R39, R41	4.7k	
R40, R42	100k	
R43, R44	39k	
R45, R46	220Ω	
R47	100Ω	
R48, R49, R50, R51	100k	
R52, R53	not fitted	(refer to the text)
R54, R55	10k	
R56, R57	220Ω	
R58, R59	22k	
R60	22Ω	
R61, R62	22k	
R63, R64	56k	
S1	Subminiature Toggle Switch on-on (panel mounting)	
S2	Subminiature Toggle Switch 2× on-on (panel mounting)	
Heatsink	140×36×19mm	
Al blade	approximately 42×50×2mm	
Push-on Knobs	small	3 pieces
Push-on Knob	average	

by high frequency interference generated in the computer. As in our case, with the amplifier installed inside the PC case, these interferences can affect the sound amplifier. The unwanted signals can get in from any undefined source through all the unprotected ways, e.g., the supply voltage, ground connections, with the input signal, or through incomplete screening. The supply voltage and the inputs and outputs should be protected by LC filters. The metal enclosure protects from the external electromagnetic field, but all this is not enough.

You can detect interfering signals that still get into the amplifier on the double ground from the direction of the supply voltage and the input port. It is not possible to prevent them by filters. The only solution that remains is the galvanic separation of the input stage. The test results are shown in *Table 1*.

BLOCK DIAGRAM

Figure 1 shows the complete block diagram. The amplifier's main line consists of the preamplifier (IC1), the tone control (IC2), and the final amplifier

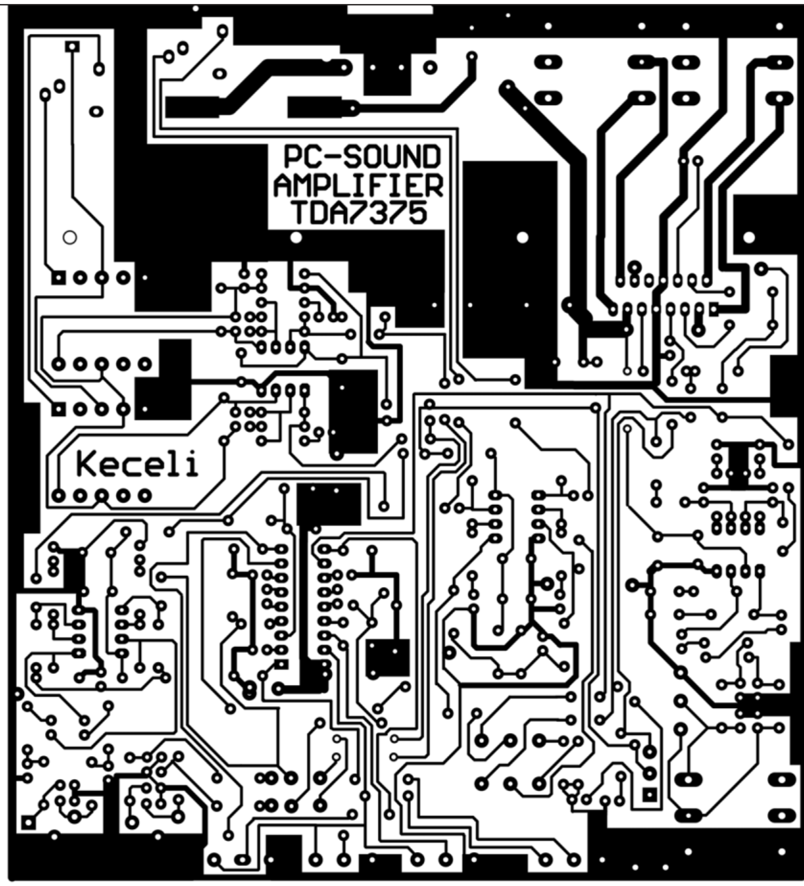

FIGURE 3: Amplifier PCB, soldering side.

(IC3). The gain of the various stages is indicated on the figure. Each stage contains an IC. With these three ICs, the amplifier can already function. However, you need to add further items to build a general-purpose amplifier. The headphone amplifier is implemented with an additional IC (IC4), and there are further two ICs: the IC6 for the REC output, and the IC5 for the PA-driver output. The PC input, the loudspeaker output, and the PA-driver output are on a separate board with individual filters. This board is built in a metal enclosure that is mounted to the back of the PC.

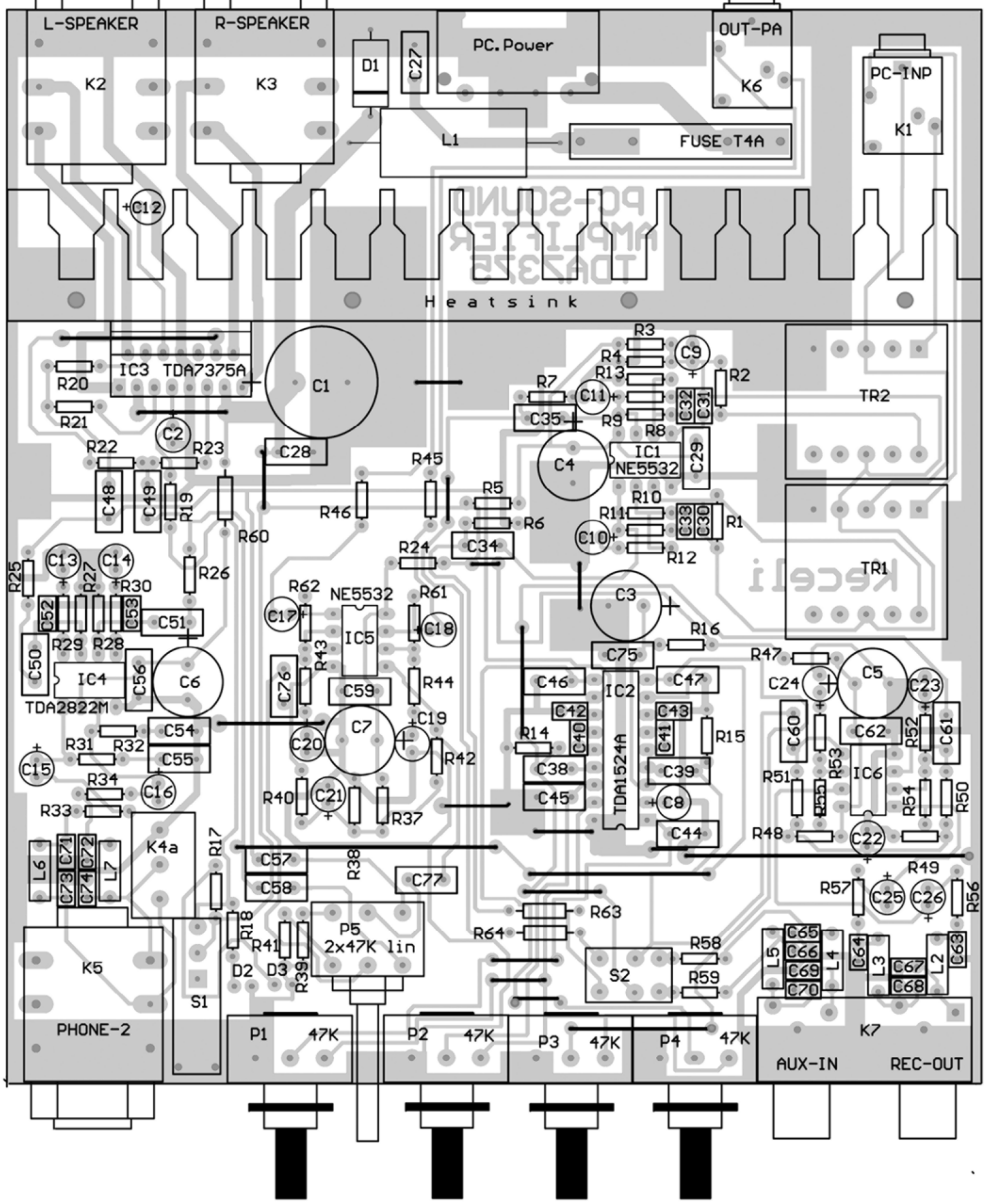

FIGURE 4: Amplifier PCB, arrangement of the components, component side.

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PRINCIPLES OF OPERATION

Figure 2 shows the amplifier's detailed schematic. The signal goes to the amplifier from the soundcard through the PC input port. This connector, which is on the back of the computer, is fully insulated and fitted on a terminal panel along with the PA-driver output and the loudspeaker connectors. Ferrite beads suppress the interfering signals.

From here, the signal goes to the amplifier through a screened cable and K1 connector (for internal wiring), which is on the amplifier panel. K1 is insulated the same way as the input connector. Full galvanic separations are achieved by the input transformers.

Because this is the sensitive point of the whole system, you must choose a good quality one-to-one transformer, which you can purchase off-the-shelf or make it at home. I used a homemade transformer.

For the sake of a better bass fidelity, I found it advisable to build in a bass correction circuitry. This is the actual pre-

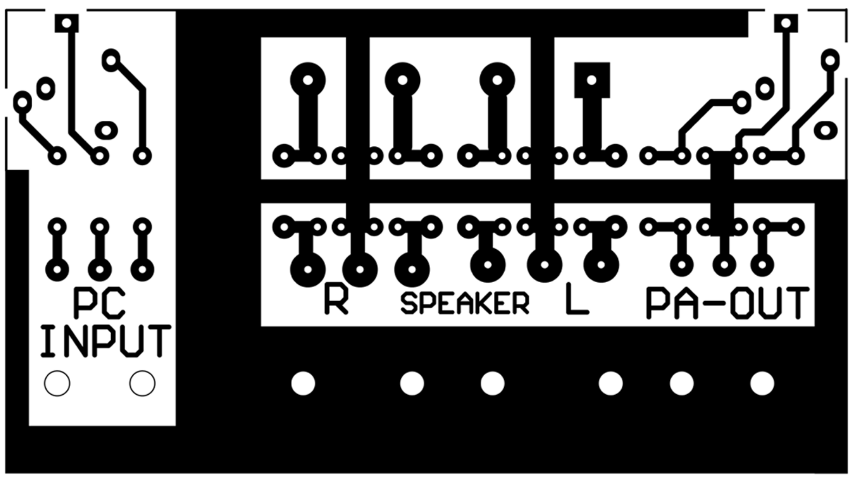

FIGURE 5: Terminal PCB, soldering side.

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amplifier IC1. The solution is a conventional low-noise OP. Its gain is hardly more than one (at 1kHz). R1 and R2 resistors determine the impedance matching, while C30 and C31 set the lower frequency roll-out.

The U_{ref} produced by the R3/R4 resistors and filtered by the C9 goes through the winding to the input of the amplifier. Bass boost for the compensation of the transformers' bandwidth is ensured by components C32-R9 (C33-R11). The signal from the amplifier goes to the input signal selector through the C34, C35 capacitors and the R63, R64 resistors. The DC null potential is produced by the R6, R7, R14, and R15. This is necessary to prevent the switchover click.

Because the gain of the other stages is quite high, it is the low frequency boost that is more characteristic of the preamplifier. The unnecessary gain is reduced by R63 and R64. If you change R63 and R64, you can alter the input sensitivity. However, increasing the gain is not necessary and can be risky because the amplifier's low-frequency stability may deteriorate.

The AUX input-RCA connector pair is

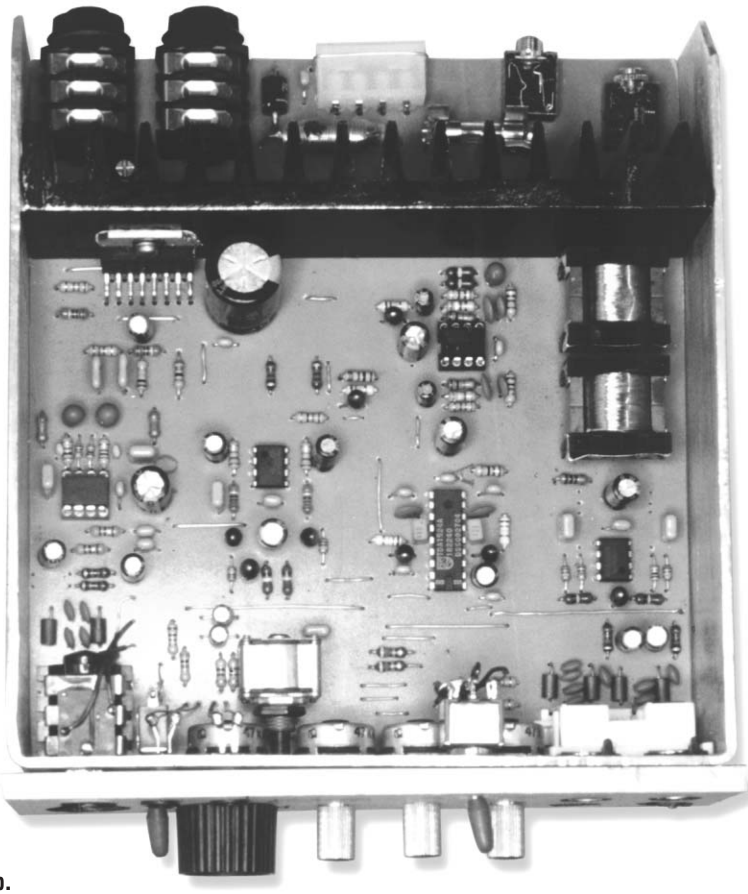


PHOTO 2: Interior of PC-amp.

installed on the front panel of the amplifier's PCB. Similar to the previous one, these connectors also contain filters made up of ferrite beads and capacitors. These items are fixed to the amplifier panel, near the terminal block.

The resistors R58 and R59 determine the sensitivity of the input. Because the 22k value is calculated for standard value, you can connect an external CD player, tuner, or other signal to it. Input selection is done by a switch, so the signal reaches the tone control stage. The tone control task is carried out by the IC2. Its gain is 20dB, which you cannot alter.

Besides the bass and treble tone control, you'll also find the volume and balance control here. This is a complex stage; control is carried out by DC levels. From the point of view of construction, this makes the stage less complicated since the RC items of the usual tone control stage are not required and mono potentiometers are sufficient for the control.

The IC2 contains a loudness tone correction stage, as well. This plays an important role by setting the sound transmission to the sensitivity of the human ear. In our case, this stage is always switched on. You can switch it off for experimenting by grounding the IC2 17 pin by 2.2k.

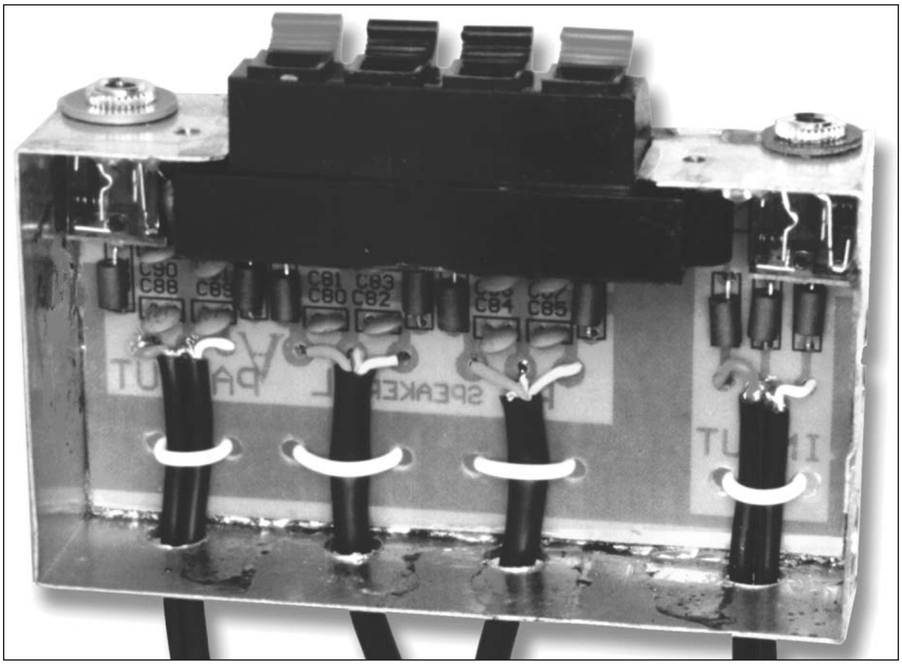


PHOTO 3: Interior of terminal block.

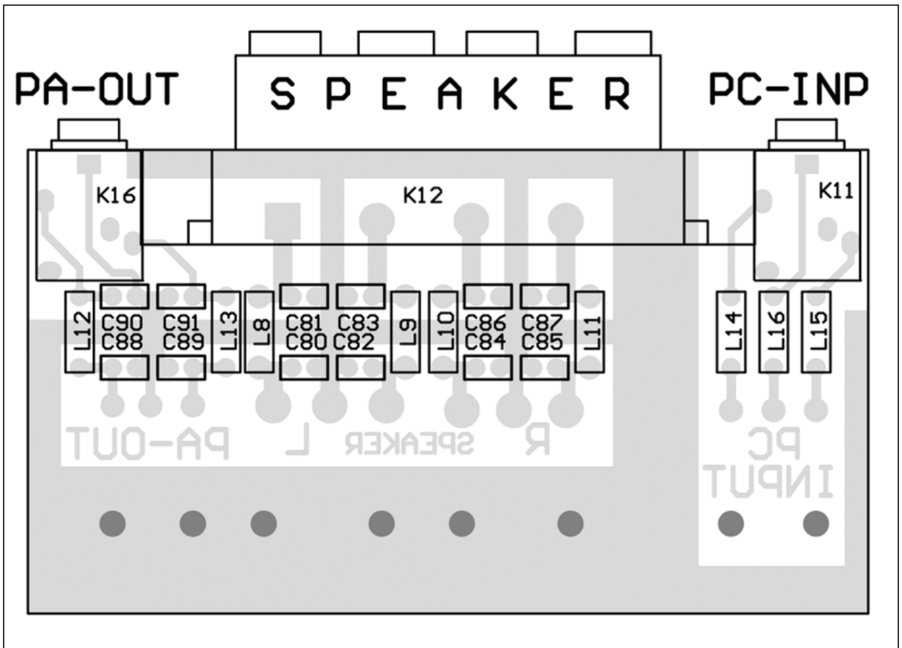


FIGURE 6: Terminal PCB, arrangement of the components, component side.

After setting the frequency response, the signal goes to the power stage through the C48 (C49) capacitors. The power amplifying is carried out by the IC3. For this, I chose an item that requires only a few external components and can still produce good quality (as you have previously amplified the signal properly). Its gain is low.

The gain of the TDA 7375 is fixed at 26dB. I reduced this to 20dB with external components. According to the manufacturer's manual, the inverted and non-inverted inputs of the IC are joint. I separated them by building in ballast resistors (R20, R21, R22, and R23).

As there is a feedback on the inverted input, this input is different from the non-inverted input, so the resistor values are different. I chose the resistor values so that they would receive identical output voltages. The inverted inputs provide phase-shifted signal, which is required for the functioning of the bridge circuit. In this way, no separate phase inverter is needed. In case of an IC with higher gain, the system might not be stable any longer and would make the operation impossible.

As already mentioned, TDA 7375 IC has a standby pin that is connected to the switch on the front panel. Shutting down the whole amplifier is not possible, but you can switch off the loudspeakers with the standby switch.

As the other stages of the amplifier operate independently of this, you can listen to the music through headphones, without external disturbance. K2 and K3 connectors are the loudspeaker connectors on the PCB, but only for internal connection, as the loudspeaker connector K12 is on the back panel on the terminal PCB. Propagation of interfering signals is hindered by a ferrite bead filter.

OUTPUT STAGE

The amplifier can already function this way, but for the sake of the general-purpose applications, I built in three more ICs. As the PA contains a standby switch and the switching over of the loudspeaker output would be more complicated, I constructed a separate amplifier for the headphone output. This is the IC4, TDA 2822, similar to the power amplifier, but with lower power IC.

According to the manufacturer's manuals, the gain is not variable. However, by increasing the negative feedback, I managed to find an optimum solution by reducing the gain using external components and optimizing a reliable state. If the gain is different from the optimum, the amplifier is apt to pop noise. The signal goes to the IC4 through the 68k resistors (R25, R26). This value is usually appropriate for the headphone, but if needed, you can change the value of the resistors.

To avoid pop noise, the previously mentioned IC requires some more external components. I added the C52, C55, and R31 (C53, C54, R32) to it for this purpose. The R33 and R34 capacitors serve as protectors of the IC and of the headphone. The output signal goes through the ferrite bead filters to the connectors mounted on the front panel and the PCB. The smaller connector is mounted above the larger one and is wired to the K4a connector that is on the PCB. This makes the universal connection of headphones possible.

The IC6 gives an output signal from the preamplifier for recording. The circuitry of the stage is conventional. The stage does not need a gain to reach 95mV, it operates only as a buffer amplifier.

Hence the places of the R52, R53, C23, and C24 components are given, but they are not installed. By mounting these components, you can achieve higher

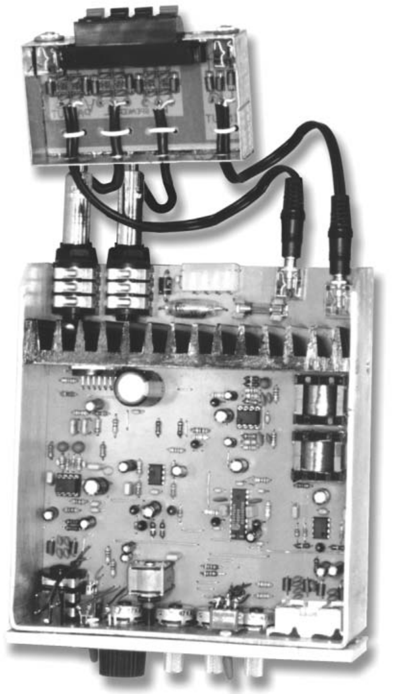


PHOTO 4: Interior of PC-amp and terminal block width interconnection.

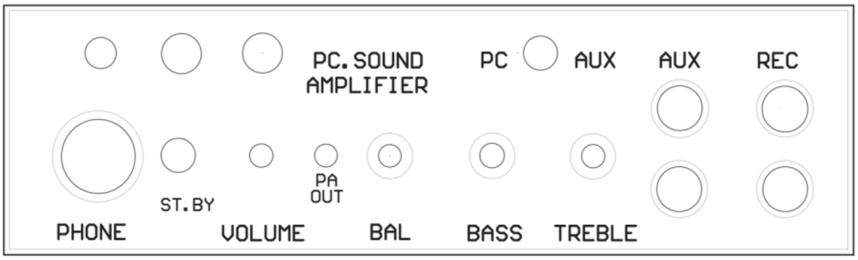


FIGURE 7: Front panel drawing.

A-2246-7

gain and output level. The output is filtered by the already described ferrite bead filters. The RCA connector pair is on the front panel of the amplifier PCB.

The IC5 is an OP amplifier, as well, with a conventional circuitry. The signal reaches this stage from the tone control IC, which means that it gets the same signal as the PA. The IC has minimal gain, and the output level is adjustable (using a screwdriver) with the P5 that is on the front panel. This output can serve as a PA-driver (or line) output. The K6 connector is on the amplifier PCB and is connected by a cable to the terminal PCB, where you'll find the filters and the K16 connector.

POWER SUPPLY AND FILTERS

The power supply connector is on the amplifier panel. The D1 diode protects from over-voltage. L1 and C27 act as filters. Furthermore, every IC has its own supply filter, which consists of a 100nF and an electrolytic-capacitor. You need an especially high value electrolytic-capacitor C1 to protect the power amplifier. The filter capacitors are directly beside the ICs. The operation amplifiers (IC1, IC5, IC6) have an asymmetrical supply, so the production of an U_{ref} is necessary.

+6V is achieved by the serial connection of two resistors. This voltage is filtered by an electrolytic capacitor and is led through resistors (except for IC1—there it is led through a transformer) to the input of the amplifiers (3.5 pin). The environment is protected by ferrite bead filters (fitted directly next to the connectors) from the high-frequency interference that is radiated by the computer. As the loudspeaker's output, PC-input and the PA-driver output connectors are placed on the computer back panel and are fitted in a separate metal enclosure on a separate PCB (terminal panel).

CIRCUIT ASSEMBLY

All the active components with the elements belonging to them coupled with the front panel connectors and the operating devices are on the very same PCB (called amplifier panel). This is built in the computer in a vacant driver space. The K1, K2, K3, and K6 connectors placed on the amplifier panel are only used for internal wiring, because further connectors are provided on the back panel. The loudspeaker connector, the PC input, and the PA-driver output with the ferrite bead filters belonging to them can be found on a separate PCB (terminal PCB) in a metal enclosure that is attached to the PC's back panel.

The amplifier PCB is in a metal box made of a 2mm Al (Aluminum) sheet and enforced with a metallic frame at the front and both sides. The control devices of the front panel are attached to this frame. The heatsink is in the middle of the PCB module to which the already mentioned frame is fixed, so that a closed ring is formed. The front panel connectors with the filters belonging to them are screened by a thin metal sheet. A front panel hides the screws of the Al frame, but this has only an esthetical function.

ASSEMBLY

Assembly of the amplifier is very simple; no adjustment is required. If you work carefully and avoid mistakes, the amplifier will work at once. You should carry out the tests using an external power supply and a signal generator. After having mounted the PA-IC3 and the tone control IC2, you can try the first test. The amplifier must function with the AUX input. After this, build in the preamplifier together with the IC1.

The other stages should not cause problems either. (In case one of the peripheral stages is not needed, you can leave it out.) Because of limited space, the smaller headphone connector is not placed on the PCB, but is mounted directly on the front panel. The S2 switch and LED are mounted to the front panel as well.

After having tested what you have built so far, it's time to install the amplifier.

Warning: while installing and connecting inside the PC, you should turn off the unit. Connecting the amplifier to an operating PC might cause a system crash due to the charging current and the high capacity of the C1.

The amplifier might function without ferrite bead filters, but you should still mount the terminal PCB on a vacant space on the back of the PC. You must install the connector K11 isolated from the metal case. The PC input receives the signal through a screened cable from the soundcard. As there is a metal sheet above and below the amplifier, this does not need to be screened separately, but in order to avoid a probable short circuit it is advisable to place an isolating sheet below the amplifier.

COMPONENTS

The heatsink—because I could not find one of appropriate size, I cut a bigger one than I needed. This way I produced the appropriate form. If necessary, you can use a thicker Al sheet.

The transformer—the simplest solution is to use a commercial off-the shelf 1:1, 600:600Ω transformer. I made this transformer myself, in this case, on an EF-20 ferrite core of N27 material, (Siemens-Matsushita Cat. # B66311-G90-X127), 2×600 turns (ϕ= 0.1mm). Between the two windings there is one aligning turn. You ground one of the terminations as screening; No metal screening is needed.

TDA7375A, STMicroelectronics, 2×37W dual/quad power amplifier for car radio. Minimal external components. Internally fixed gain (26dB). No audible pop during standby operations. Protections, output AC/DC short circuit, overrating chip temperature with soft thermal limiter. Supply voltage range 8 to 18V.

NE5532, Texas Instruments. Dual low-noise, high-performance operational amplifiers. They feature very low noise, high output-drive capability, low distortion, input-protection diodes, and output short-circuit protection. These operational amplifiers are compensated internally for unity-gain operation. Wide supply-voltage range . . . ±3V to ±20V.

TDA 1524A, Philips Semiconductors. Stereo-tone/volume control circuit. The device is designed as an active stereo-tone/volume control for car radios, TV receivers, and mains-fed equipment. It includes functions for bass and treble control, volume control with built-in contour (can be switched off), and balance. All these functions can be controlled by DC voltage or by single linear potentiometers. Few external components are necessary. Low noise due to internal gain. Supply voltage 7.5 to 16V. Maximum gain of volume 21dB. The input resistance is a function of the gain of the volume control. At maximum volume it is 10k, and it increases with the reduction of the gain.

TDA2822M, SGS-Thomson, Dual low-voltage power amplifier. It is intended for use as dual audio power amplifier in portable cassette players and radios. Supply voltage 1.8 to 15V, output power 300mW at 32R. Closed loop voltage gain 39dB. Input resistance 100k. ❖

If you would like to purchase these PCBs from the author, or are looking for help, you may contact him at

Keceli Jenö
PF1235, 6701
Szeged, Hungary
E-mail: keczeli@yahoo.de

REFERENCES

STMicroelectronics specification publication TDA 7375A, http//www.st.com.

Texas Instruments production data information NE5532, SLOS075G – 11/79, Texas Instruments, PO Box 655303, Dallas, TX 75265.

Philips Semiconductors product specification TDA 1524A data sheet.

SGS THOMSON MICROELECTRONICS, TDA2822M applications.

ELV Journal (German) 4/95 PC-Audioverstärker.

ELV Journal (German) 5/99 PC-Audio-Verstärker PCA 200.

Elektor (German) 12/2000 PC-SoundAmp.

Minimal Reactance Power Supply

The author introduces a new concept for designing a current regulated power supply for your next tube amplifier or preamplifier project.

By David Davenport

I finished developing my SET line stage, and now it was time to produce a power supply. I used a standard choke-input power supply for most of the development that you read about in "Odyssey of a Line Stage" (*audioXpress* 2/03 and 3/03). That supply was okay, but I wanted something special to support the superb sonic qualities of the line stage. If you remember, the line stage ended up as a parafeed design using a constant-current source (CCS).

Thinking about power-supply topologies, I began to wonder whether it needed a choke if the load drew a constant current. This thought opened the door and allowed me to get out of the box of conventional power-supply thinking. The thoughts poured in.

In my experience a CCS was better than a choke as a plate load. Was the CCS an ideal choke? Of course, a CCS doesn't store energy like a choke does, but so what? What if I were to use a CCS in place of a choke in a power supply?

Figure 1 shows the concept with CCS1 in the power supply and CCS2 in the line stage. In this example CCS1 provides a constant 35mA, CCS2 draws a constant 30mA, with the remaining 5mA flowing through the resistor, which is chosen to produce the desired voltage on the output of the power supply. I built a model of the design shown in *Fig. 2*, and it worked just as I expected.

CAPACITOR VALUE

My next question was "How big does the input capacitor need to be?" One of the important characteristics of a CCS

PHOTO 1: The completed supply.

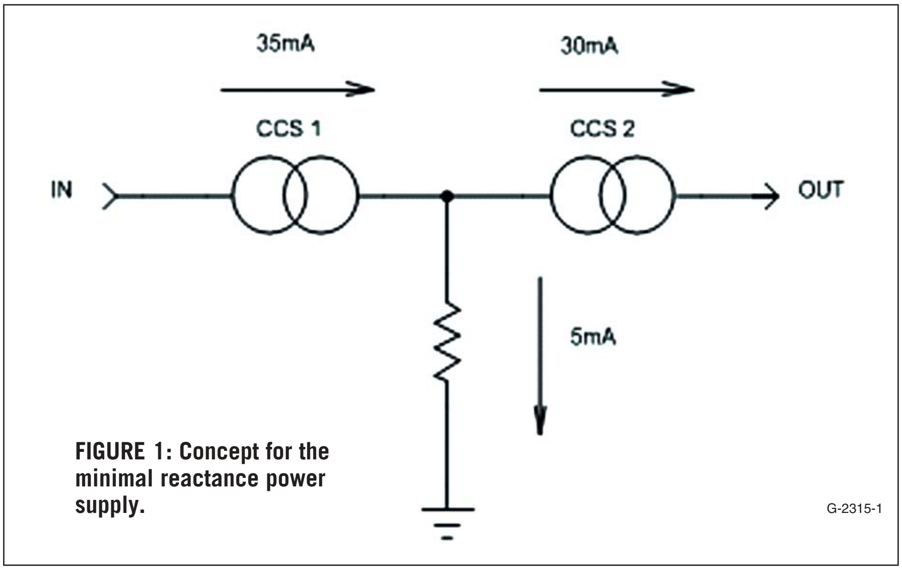


FIGURE 1: Concept for the minimal reactance power supply.

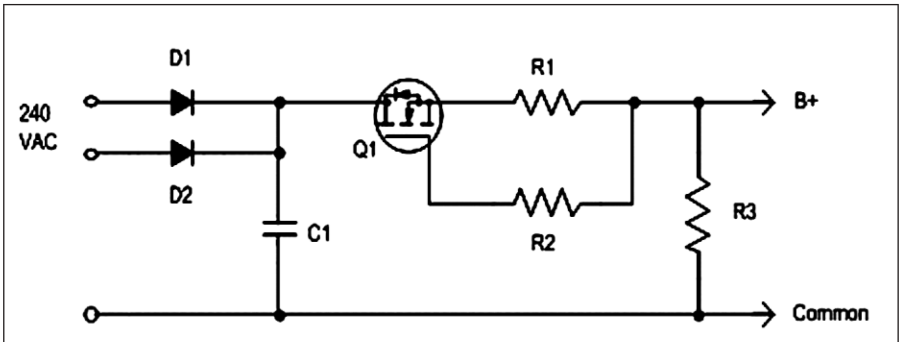


FIGURE 2: Simple prototype.

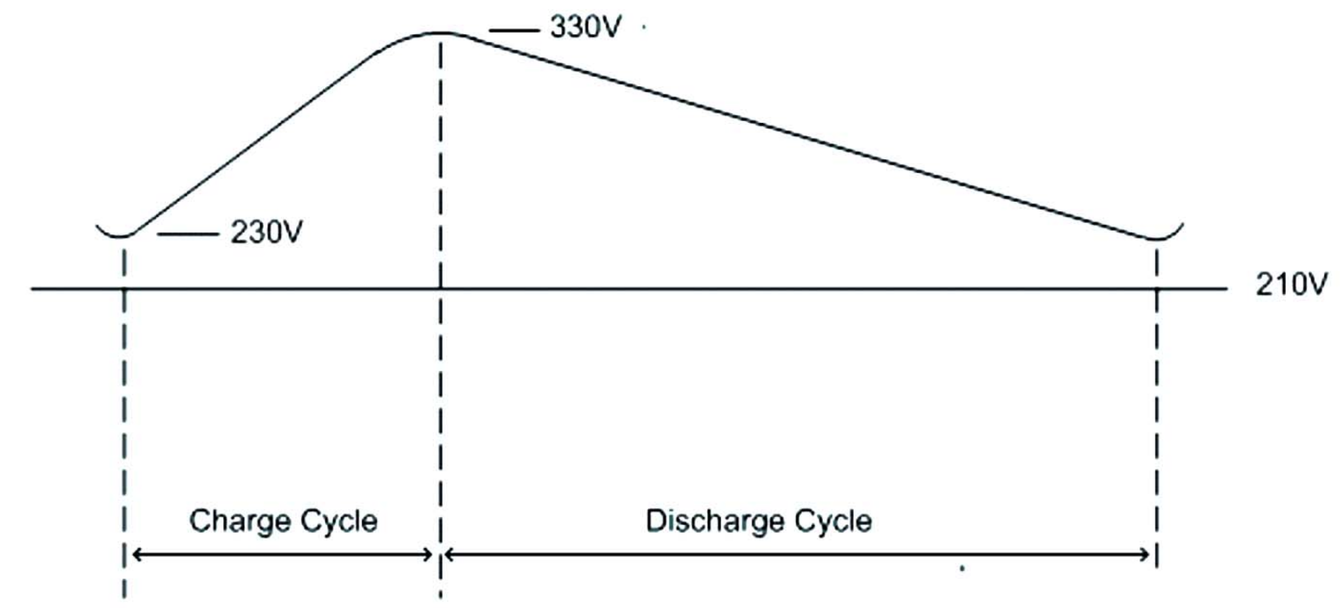


FIGURE 3: Filter capacitor ripple. G-2315-3

is its power-supply ripple rejection (PSRR). With a very high PSRR, the CCS can tolerate a large ripple on its input; in fact, the amount of ripple is not important as long as the instantaneous voltage stays above a certain value.

In order for the CCS to provide a constant current, there must be sufficient voltage on its input. If the voltage were to drop below this point, the CCS would act like a large resistor and the current would drop as the voltage dropped. It doesn't matter whether the voltage rises above the critical value; the output current remains constant.

Although the DN2450 high-voltage MOSFET I used here needs only a few volts across it to work properly, I conservatively allowed 20V. If I wanted 200V on the output, all I would need to do is keep the input above 220V. I could have a hundred volts of ripple and, as long as the instantaneous voltage stayed above 220V, the CCS would reject all of the ripple. This means that the input capacitor does not need to be very large at all, perhaps only a couple of microfarads.

But don't you need a large capacitor to store lots of energy to provide for powerful bass and a large crescendo? Nope! Isn't "the more the better" a good rule for capacitors? Nope! As long as the CCS input voltage stays above the critical value, the output is the same. The current is constant even for those deep bass thumps and large crescendos.

In fact, a large capacitor is not neces-

ABOUT THE AUTHOR

David Davenport has been involved in audio DIY for over 40 years. He retired after 30 years as an engineer with IBM, and then founded his own company, Raleigh Audio, dedicated to exploring the limits of audio electronics and providing the results to OEMs and serious hobbyists. He has written many articles for *audioXpress*, as well as the former Audio Amateur publications, *TAA*, *GA*, and *SB*.

sarily a good thing. Take a look at the ripple waveform in *Fig. 3*. During the first part of the cycle, the capacitor charges to the output voltage of the rectifier, while during the second part of the cycle the capacitor discharges until the rectifier voltage again rises to exceed the discharged voltage. As the value of the capacitor increases, the capacitor can store more energy so it loses less voltage during the discharge part of the cycle. As a result, the charge time becomes shorter and the discharge time longer.

With a very large capacitor the discharge time can be minuscule. However, since the capacitor is large, it provides low impedance during this short charge time resulting in a large amplitude current pulse. Thus, the current drawn from the transformer is in the form of very short, high–amplitude current pulses, which cause the transformer to ring like a bell.

A smaller capacitor that draws a wider, lower amplitude current pulse

TABLE 1

PARTS LIST FOR *FIGURE 4*

PART	DESCRIPTION	SOURCE
Resistors		
R1, R2	0.5Ω, 5W	Mouser 71–RS5–0.5
R3, R4	0.47Ω, 7W	Mouser 280–PRM7–0.47
R5	120k, ¼W	Mouser 271–120K
R6	100k, ¼W	Mouser 271–100K
R7	100Ω, ¼W	Mouser 271–100
R8	40.2Ω, Caddock MK132	Michael Percy
R9	10k, 5W Mills MRB–5	Michael Percy
R10	40.2k, 12W Mills MRB–12	Michael Percy
R11	5k trimpot	Mouser 72–T93YB–5K
Capacitors		
C1	0.47µF, 300V X–rate	Digi-Key P11120–ND
C2	100nF 50V	Digi-Key P4923–ND
C3	3.6µF, 600V Solen	Michael Percy
C4	1µF, 600V Solen	Michael Percy
C5	100nF, 630V	Mouser 75-MKP1840410634
C6	10nF, 1600V	Mouser 75-MKP1841310134
C7	6800µF, 35V	Digi-Key P6660–ND
C8	1µF, 50V	Digi-Key P5174–ND
C9	220µF, 10V	Digi-Key P10220–ND
Inductors		
L1	CM choke	Digi-Key PLK1007–ND
L2, L3	EMI bead	Digi-Key P9817BK–ND
Diodes		
D1, D2	FRED, DSEI–12	Digi-Key DSEI12–12A–ND
D3	1N4005	Digi-Key 1N4005GICT–ND
D4-D7	SB540	Digi-Key SB540CT–ND
Semiconductors		
Q1	MOSFET, DN2540	K&K Audio
U1	Regulator, LM78M06	Digi-Key NJM78M07FA–ND
U2	Timer, LM555	Digi-Key LM555CNNS–ND
Miscellaneous		
T1	Transformer, Lundahl LL1683	K&K Audio
V1	12×4	Antique Electronics
K1	Relay	Digi-Key PB295–ND
	7-pin tube socket	Antique Electronics
	Heatsink	Digi-Key HS241–ND
	Insulator, heatsink	Digi-Key BER102–ND
	Insulating bushing	Digi-Key 3049K–ND
	4–40 screw & nut	Procure locally
	Mounting hardware	Procure locally

Note: A kit is available from K&K Audio

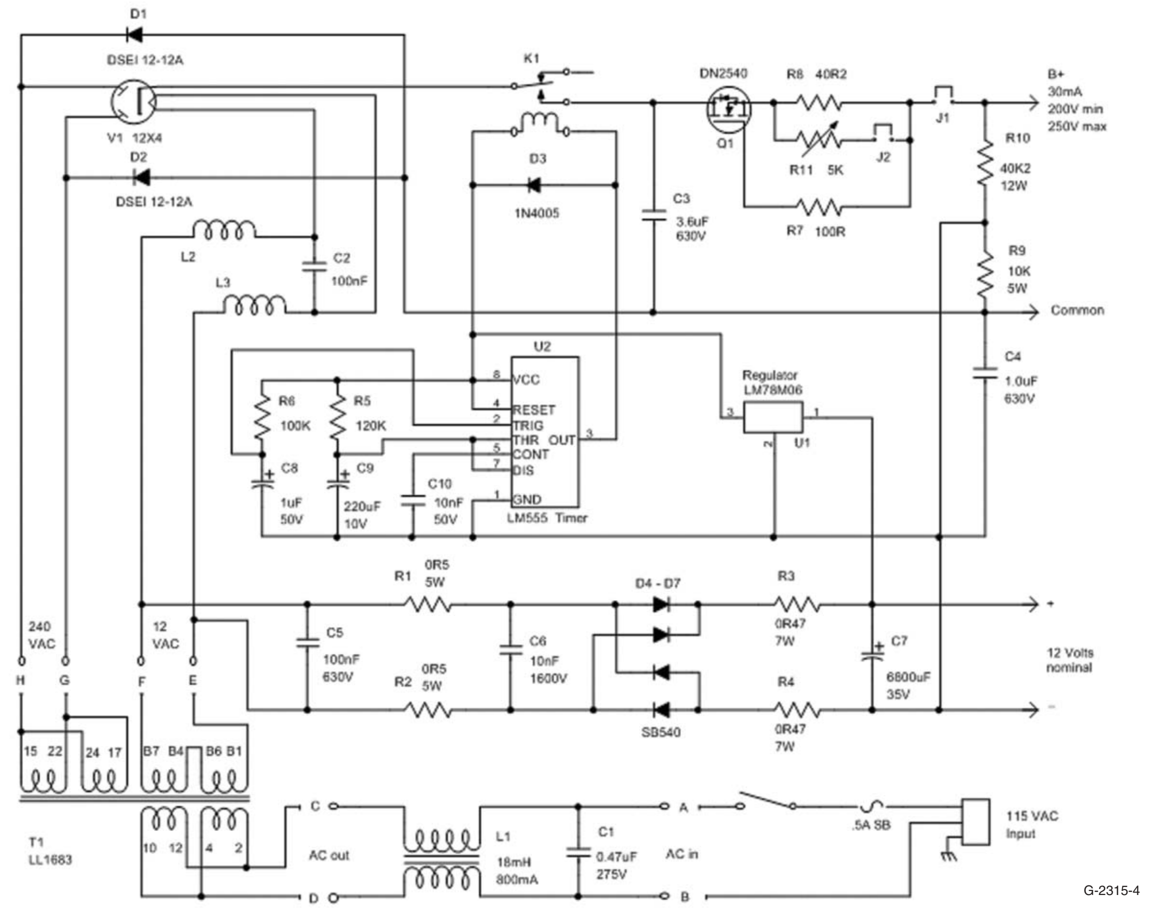


FIGURE 4: The complete power supply.

has the advantage here. Another benefit of using a small capacitor is that it can be a high–quality film rather than an electrolytic, resulting in even better sonic quality.

The result of all this is a unique supply topology that regulates the current and lets the voltage seek whatever level is needed to provide the required current. Because the design uses as small a capacitor as needed and does not incorporate an inductor, I call it a minimal reactance power supply.

DESIGN STEPS

Take these steps to design your own minimal reactance power supply:

1. Determine what current and voltage the external load needs.
2. Calculate the current through the load resistor (R3 in *Fig. 2*) that will develop the desired output voltage, E = IR.
3. Calculate the required regulator current. This is the sum of the current through the resistor that you just calculated plus the current required by the external load.

4. Calculate the size of the filter capacitor (C1) needed. For a rule of thumb, I use the formula: C = 10 I/V, where C is in microfarads, I is the current through the CCS in mA, and V is the peak-to-peak voltage of the input ripple. The top peak is the maximum output of the rectifier, and the bottom peak is 20V greater than the desired B+ voltage. So, for example, if I wanted 210V B+, 35mA through the CCS, and my rectifier put out 330V, the capacitor needed would be $C = 10 \times 35/(330\text{-}230) = 3.5\mu F$. Use the next largest standard value available.
5. Install the capacitor in the C1 location.
6. R2 is a gate stopper, which will curb any tendency for the CCS to oscillate. The value is not critical; 100Ω should work fine here.
7. Determine the value for the current sense resistor (R1).
   a) Insert a variable resistor into the R1 position.
   b) Connect a milliamp meter in series with R1.
   c) Connect the actual external load (or a resistor that will draw the same amount of current) to the output of the supply.
   d) Apply input power and allow enough time for any tubes to warm up.
   e) Adjust R1 for the desired current through the milliamp meter.
   f) Check that the residual current through R3 produces the desired voltage.

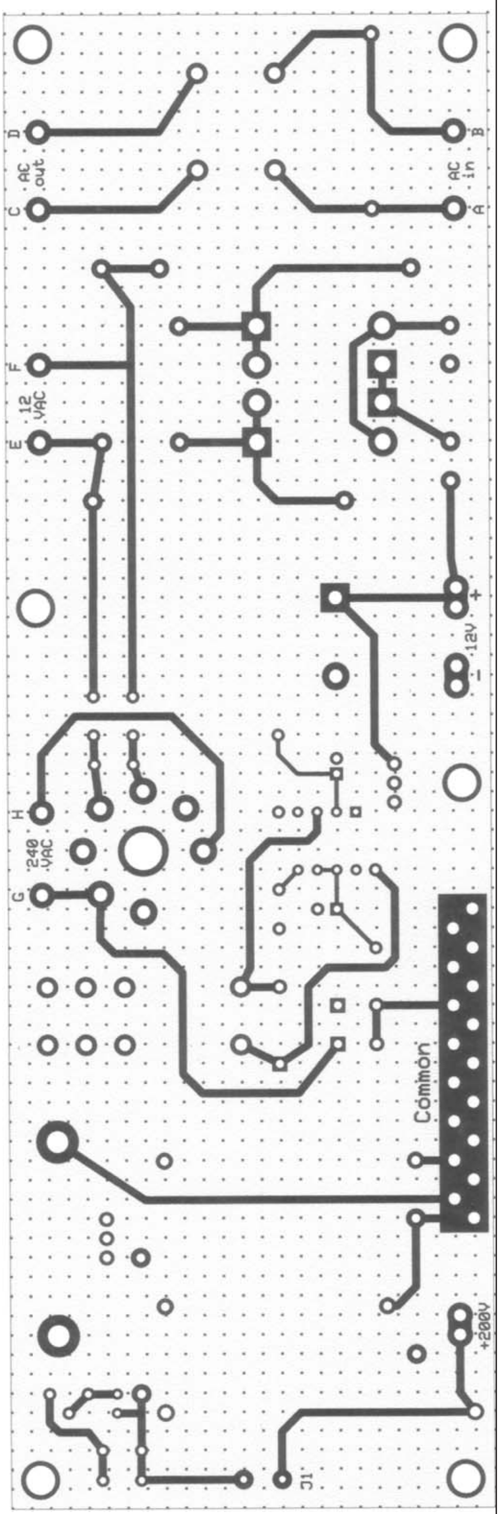


FIGURE 5: Top layer of the printed circuit board. G-2315-5

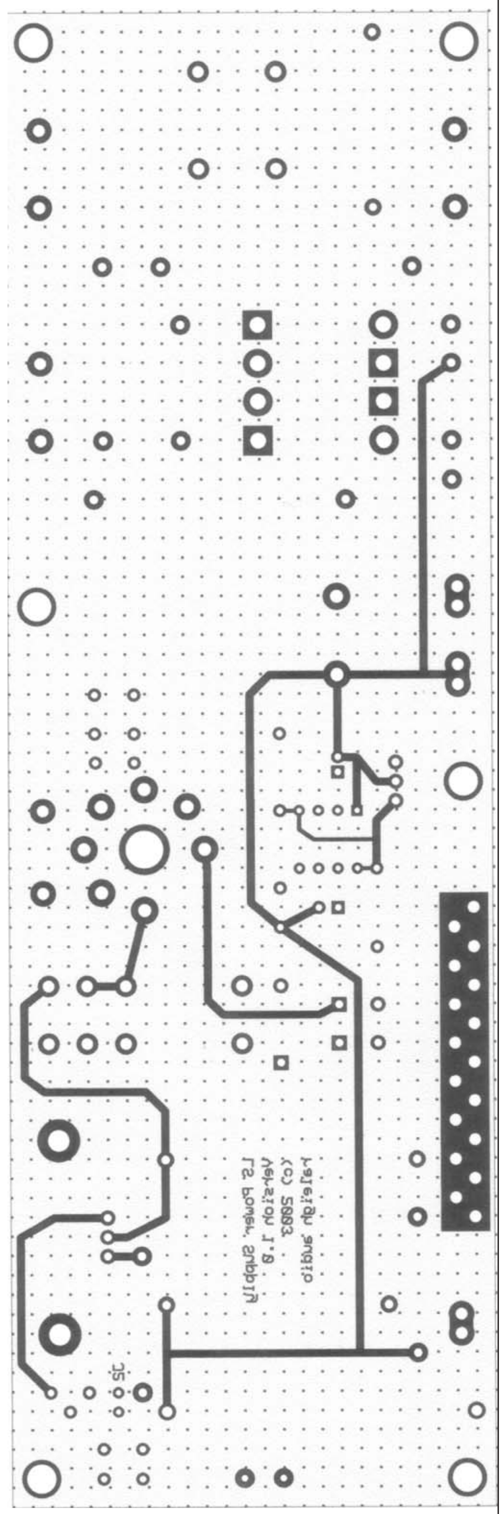


FIGURE 6: Bottom layer of the printed circuit board. G-2315-6

SOURCES

Antique Electronic Supply
6221 S. Maple Ave.
Tempe, AZ 85283
480-820-5411
FAX 480-820-4643

Digi–Key
PO Box 677
701 Brooks Ave. S.
Thief River Falls, MN 56701-0677
218-681-3380
FAX 218-681-3380

K&K Audio
1004 Olive Chapel Rd.
Apex, NC 27502
919-387-0911

Michael Percy Audio Products
11731 Stillwater Creek Rd.
Nevada City, CA 95959
530-470-8650
FAX 530-470-8651

Mouser Electronics
1000N. Main St.
Mansfield, TX 76063
800-346-6873
FAX 817-804-3899
sales@mouser.com
www.mouser.com

If not, adjust the value of R3 or the current through the CCS.

g) Measure the value of the variable resistor and replace with a fixed resistor.

This is all there is to the basic concept; however, your final design may have some refinements. This supply is transparent, so it is important to use high-quality components. Let me share the complete design of the line stage supply as shown in *Fig. 4*.

FINAL DESIGN

The first thing you will notice is that I have added a power-on-delay timer. Since the supply needs the load to participate in drawing current, it is important that the load be capable of drawing that current. The delay gives the tubes in the line stage a chance to warm up before applying the voltage to them.

Rectifier tube V1, along with diodes D1 and D2, form a hybrid Graetz bridge, which displays the audible characteristics of an all-tube rectifier. Because of the resistance of the tube in series with the solid-state diodes, there is no need for a reverse recovery spike filter. The 50k output resistor is split to provide a bias to the 12V filament supply. Diodes D4 through D7, together with their associated components, form the 12V supply used to power the timer as well as the line stage filaments.

Common mode choke L1 along with capacitor C1 constitute a power line filter, and the high-quality Lundahl LL1683 is used as a power transformer. *Photo 1* shows the completed supply, and printed circuit board layouts are shown in *Figs. 5* and *6*. The parts list is in *Table 1*.

Although this power supply is specifically intended to be used with a load that draws a constant current, you could easily adapt it to use with any load. All that you would need is to add a shunt regulator in place of the load resistor. This regulator could incorporate a comparator or be as simple as a single VR tube.

Build a minimal reactance supply for your next audio project, and I believe you will be extremely pleased with the clean, quiet power it supplies and the sonic results it provides. ❖

Speaker Cables Review

By Charles Hansen and Muse Kastanovich

The parameters of any cable are defined as a combination of electrical resistance, capacitance, and inductance. The DC resistance is a function of conductor length and cross-sectional area. It does not matter whether the total area is composed of a single conductor or multiple conductors.

As the frequency increases, the resistance of a conductor increases due to skin-effect losses. This is the phenomenon by which the alternating current in a conductor concentrates near the surface, increasing the effective resistance. Round conductors show a more pronounced skin effect than thin flat conductors.

CABLE THEORY

The voltage between two conductors produces an electric field. When an insulator is interposed between the conductors, the molecules of the insulator rotate slightly and produce a field within the insulating material that opposes the applied field. The electric field E is reduced below the equivalent free space applied field intensity E_0 by a factor equal to the insulating material's dielectric constant relative to free space, ε_r.

$$E = E_0 / \varepsilon_r$$

The lower the dielectric constant of the insulator, the less the reduction (loss) in the total electric field.

Flat wire cables form parallel plate capacitors. The capacitance can be calculated from:

$$C = (\varepsilon_0 \varepsilon_r A) / d$$

where ε_0 is the permittivity of free space, and ε_r is the relative dielectric constant of the insulating material. This varies with material, temperature, and frequency. A is the area of a plate, and d is the separation between the plates.

The round wire cables form parallel wire capacitors. You can calculate the capacitance of this arrangement without a ground plane by:

$$C = \pi \varepsilon_0 \varepsilon_r l / \log_e (d/r)$$

where e = 2.718, ε_0 is the permittivity of free space, ε_r is the relative dielectric constant of the insulating material, l is the length, d is the separation between the wires, and r is the wire radius. Note that d must be much larger than r, a condition not always met with speaker cables. Large gauge, closely spaced round conductors may more closely approximate parallel plate capacitors.

Speaker cables also have the properties of series inductors. You can determine the inductance of two rectangular strips by:

$$L=[0.4l(0.5 \log_e (2dl/(w+h)) + 0.22(w+h)/l]$$
$$\text{where } l \gg w, h$$

where l is the length, d is the distance between the strips, w is the width of a strip, and h is the height of a strip.

To find the inductance for two parallel wires carrying equal and opposite currents, use:

$$L = [0.1 + 0.4 \log_e (d/r)]\, l$$
$$\text{where } (d \gg r)$$

where d is the distance between the

PHOTO 1: Alpha-Core Goertz MI 2.

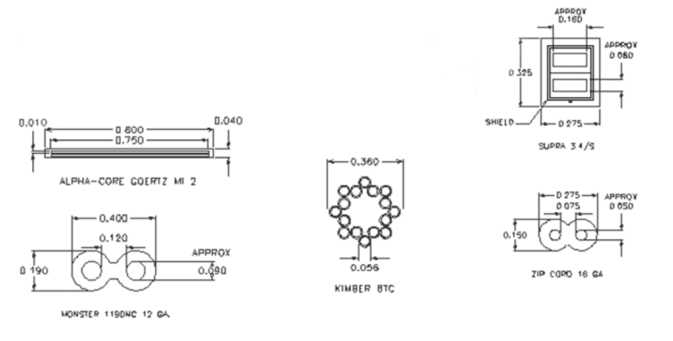


FIGURE 1: Cable cross-section and dimensions. A-2276-1

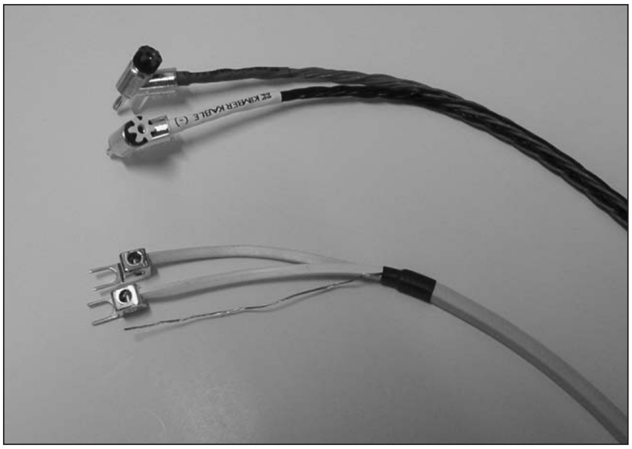


PHOTO 2: Kimber Kable 8TC and Supra 3.4/S.

wires, r is the wire radius, and *l* is the length.

Thus you can see that the impedance of a speaker cable will vary with frequency as well as the distance between conductors. Capacitance varies in proportion to distance, while inductance varies inversely with distance.

The wire resistance is in series between the amplifier and loudspeaker, and increases with frequency due to skin effect. The skin effect is essentially the same for a conventionally stranded wire as it is for a solid conductor of the same material and the same net cross-sectional area. The skin depth is the depth below the conductor surface where the current density has decreased by 1/e, and for copper at 20kHz it is about 0.5mm (0.02″).

Skin effect at 20kHz will increase the resistance by 7% to 34%, depending on the cross section profile. This presents a problem in measuring impedance at higher frequencies. As Bob Pease of National Semiconductor pointed out, when you try to measure pure inductance, you may also measure some of the skin effect. Cable capacitance tends to cancel inductance, so you may confuse inductive phase shift measurements with skin effect.

The cable inductance is also in series between the amplifier and loudspeaker. The inductive reactance in ohms is equal to $2\pi fL$. The resistance and reactance do not add arithmetically because the reactance lags resistance by $-90°$, so the series impedance is the root-sum-square of the two.

Capacitance has the least effect on the performance of a speaker cable. This is because the cable capacitance is small to begin with and is shunted by the significantly lower amplifier and loudspeaker impedances. Capacitive reactance in ohms is equal to $1/(2\pi fC)$. Capacitive reactance leads resistance by $+90°$. The shunt capacitance will tend to cancel some of the series inductance as frequency increases.

The complex sum of these L, R, and C components determines the equivalent series impedance of a speaker cable. The phase shift introduced by a cable is proportional to its complex series impedance. The flat wire cables have higher capacitance but lower inductance than the round wire cables.

CABLE CONSTRUCTION DETAILS

There are seemingly endless processes for constructing speaker cables. The cables provided for testing include a flat-wire type, one with woven wires, one with stranded wire in a rectangular jacket, and one conventional round stranded wire cable. I also added a sample: the infamous 16-gauge zip cord. Test sample lengths varied from 3′ to 10′. The 3′ lengths may not be long enough for systems with satellites on stands. I used a vernier caliper to measure the dimensions of all the cables. The results are shown in *Fig. 1.*

The Alpha-Core Goertz MI 2 "T"-series is a bi-wire speaker cable. Each conductor is a 0.35 × 0.010″ solid band of high-purity oxygen-free copper (OFC) surrounded by a 0.005-inch (5 mil)-thick sheath of 0.8 × 0.04″ polyethylene terephthalate (PET, or polyester, same polymer as Mylar™) insulation. The stiff conductors are placed in close mutual contact inside a 5 mil clear PET jacket. PET has a fairly stable dielectric constant of about 3.2, which decreases about 3% at 1MHz. Polyester film is slightly hygroscopic—it absorbs water by a factor of 0.2% in the presence of high humidity.

The mutual distance between the conductors is 0.003 inches, and the overall cross-section is approximately

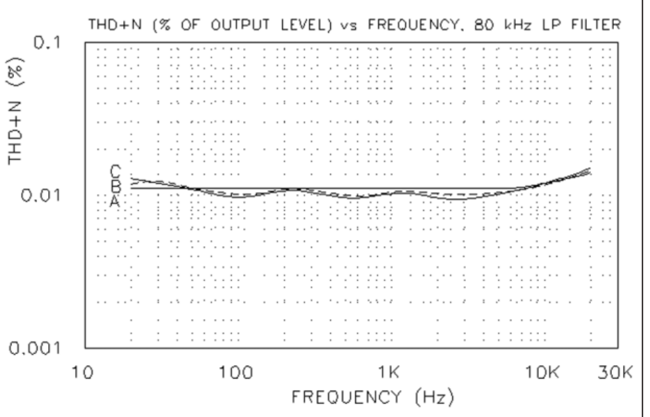


FIGURE 2: THD+N versus frequency, 3′ Goertz MI 2. A-2276-2

0.8 by 0.040″. The space ratio is close to 50%. The conductor cross-section approximates AWG 10. Rhodium-plated connectors are available to terminate the cable. *Photo 1* shows the amplifier end of the MI 2 cable test sample.

R/C Match Links are provided with the MI 2 cables. The Match Links consist of a pair of Zobel networks, 100nF in series with 10Ω. They are intended to be placed across the speaker terminals in systems with high GBW solid-state amplifiers and loudspeakers with high impedance at high frequencies to prevent oscillation. “Chances are you’ll not need these,” it says on the package.

The Kimber 8TC consists of 16 individual VariStrand™ PTFE (Teflon™) insulated wires (eight wires for each channel) arranged in a hollow, large format braid. PTFE is the most stable of the dielectric materials, with a low dielectric constant of 2.1. The aggregate wire gauge size is equivalent to two nine-gauge conductors. The tightly braided configuration and the Teflon™ insulation make these cables a bit stiff.

Various terminations are available: spades, angled and straight banana plugs, even plain ends. The test cables had WBT–0144 angled banana plugs. *Photo 2* shows the Kimber 8TC (top) and the Supra 3.4/S (bottom).

The Supra 3.4/S uses tin-plated copper wire strands arranged in a rectangular cross-section. The slash-S designation identifies the screened ply by which the two main conductors are surrounded by a braided shield with a drain wire at the amplifier end of the cable. This drain wire is intended to be connected to the amplifier chassis, to limit the EMI radiated from the cables. The shield and the ion-stabilized polyvinyl chloride (PVC) double jacketing make this cable rather stiff. The terminations are Supra Fork XL gold-plated spade plugs.

The conductors in the Monster 1190MC 12-gauge speaker cable are concentric rope lay stranded wire with eight individual helical wound members. The rope lay construction makes this cable very flexible. The insulation material is

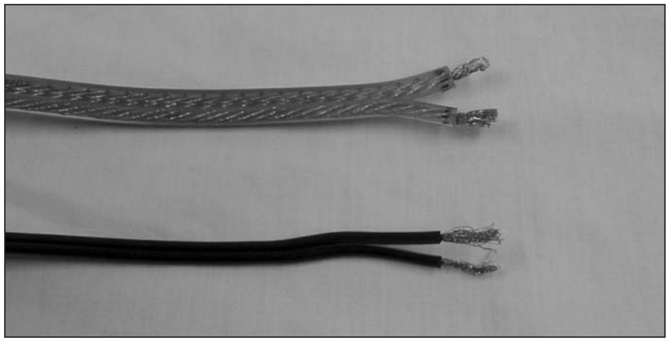


PHOTO 3: Monster 1190MC cable and 16-gauge zip cord.

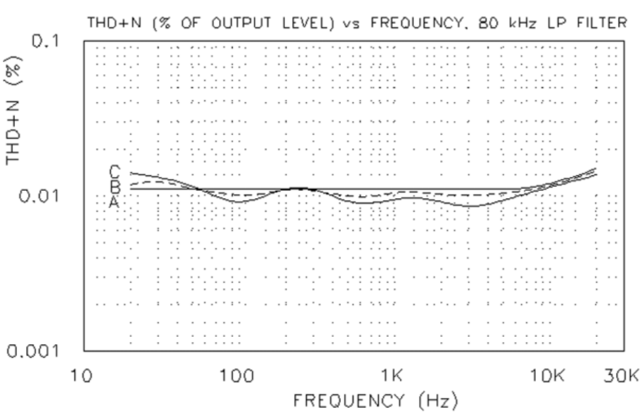


FIGURE 3: THD+N versus frequency, 8′ Kimber 8TC. A-2276-3

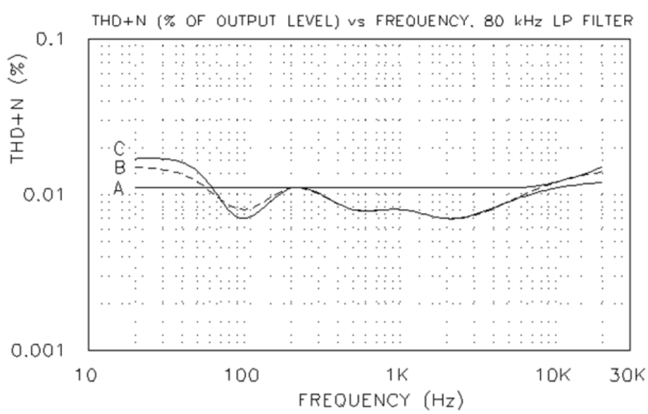


FIGURE 5: THD+N versus frequency, 10′ Monster 1190MC. A-2276-5

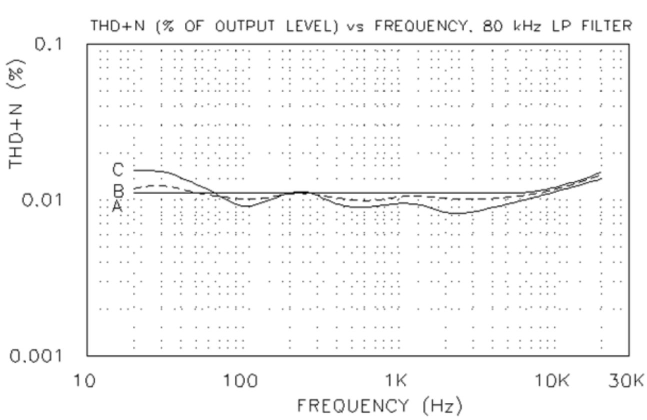


FIGURE 4: THD+N versus frequency, 8′ Supra 3.4/S. A-2276-4

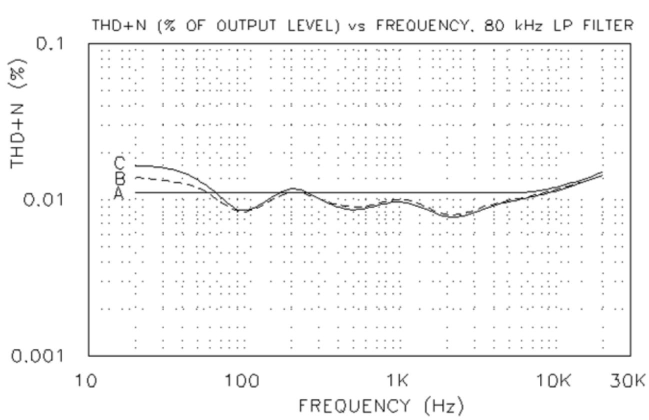


FIGURE 6: THD+N versus frequency, 8′ zip cord 16. A-2276-6

linear polyethylene, which has a stable dielectric constant of about 2.3, changing less than 1% at 1MHz. The 1190MC is Monster's original cable design.

The Radio Shack Megacable and Phoenix Gold cables appear to be similar in construction. My unofficial wire count is that each of the eight helical members contains 33 strands of AWG-37 wire. These individual strands are not insulated, so this construction is not the same as Litz wire[1]. *Photo 3* shows the Monster cable (top) and the zip cord (bottom).

The 16-gauge zip cord is intended for lower current household appliances. My sample consisted of 40 strands of 32-gauge wire, with plain PVC insulation. Ordinary PVC has a rather high dielectric coefficient of 4.7 compared with other polymers. The dielectric coefficient also decreases by 46% at 1MHz.

MEASUREMENTS

The cable samples vary from three feet to ten feet long. I measured the total capacitance, inductance, and resistance of each sample with two types of digital LCR meters. The HP 4261A performs measurements at 120Hz and 1kHz. The AADE L/C IIB uses a resonance measurement method with a variable frequency signal.

I also measured the DC leakage current of the dielectric using a Keithley 480 picoammeter as a measure of insulation quality. A 12V DC gel cell battery biased the cables. I made all measurements with the samples laid out in a straight line.

The MI 2 is quite stiff, so I needed to clamp it down to hold it flat and straight. Capacitance is measured between the two open circuit conductors (four in the case of the MI 2). The HP 4261A also measures capacitance dissipation factor (DF), which is an expression of the power loss in the dielectric. DF is the ratio of the equivalent series resistance (ESR) to the capacitive reactance (Xc), and is a measure of the insulator's ability to store energy at a given frequency. A lower dissipation factor is better.

The capacitance of the Supra 3.4/S dropped from 944pF to 328pF when I attached the shield drain wire to the guard terminal of the HP 4261A. This is because an adjacent ground plane reduces the mutual capacitance. Were the ground plane to be in between the conductors (a Faraday shield), the capacitance drop would be even greater.

Inductance is measured by short-circuiting one end of the cable and connecting the LCR meter to the open end. Resistance is measured the same way, so the L and R per foot is the sum of both conductors. The inductance measurements for the MI 2 were a bit erratic because of the bi-wire pair parallel cables. I ended up shorting the amplifier end, then measuring each of the pairs independently. Then I computed the equivalent parallel inductance Lp=La*Lb/(La+Lb).

This measured inductance of 18nH/ft for the MI 2 turned out to be higher than the specified 6nH/ft. This caused the calculated characteristic impedance, which is equal to √(L/C) and is independent of length, to also be higher than specified.

I also took a 10′ length of the flexible Monster 1190MC wire and measured its parameters first as a straight run of cable, and then coiled its full length on the floor in 8″ diameter loops. The total capacitance increased by 7% while the

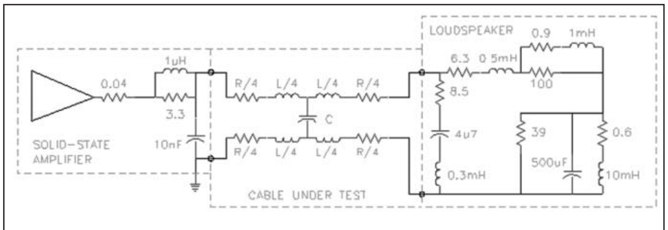


FIGURE 7: Solid-state amplifier system SPICE model. A-2276-7

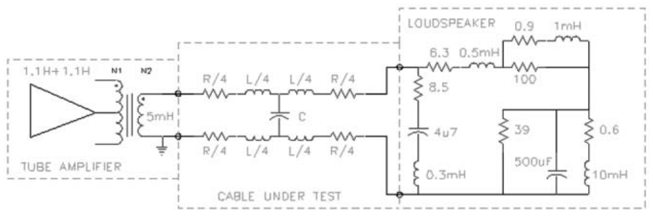


FIGURE 8: Tube amplifier system SPICE model. A-2276-8

inductance remained the same.

The standard 25°C resistance for one-foot pairs of wires is 1.58mΩ for 9-gauge, 2mΩ for 10-gauge, 3.18mΩ for 12-gauge, and 8.03mΩ for 16- gauge.

Using the measured data, I calculated the C, L, and R per foot, the characteristic impedance (Zo), the insulation resistance (Rp), and the cable resonant frequency. Using the physical measurements of the cables, I calculated Rac/Rdc ratio and the effective skin effect resistance at 20kHz. This varied from 107% of the DC resistance (Goertz) to 132% (zip cord). The data is summarized in *Table 1*.

The Goertz, Kimber, and Supra cables came with characteristic data. I compare this specification data with my measured data in *Table 2*.

TESTS

With my audio system warmed up, I first measured the THD+N versus frequency with 2V RMS across an 8Ω resistor connected directly to my Parasound HCA-1000A solid-state amplifier output terminals. Next, I connected each of the test cables in series with the 8Ω resistor and measured the THD+N at the resistor. There was essentially no difference in the readings with or without the cable. Unfortunately, I did not have a tube power amplifier available for these tests.

I repeated the THD+N vs. frequency test with each cable connected between the amplifier and one of my satellite loudspeakers, an NHT Super-One. I held 2V RMS at the amplifier terminals. Now there was some measurable difference compared to the THD+N that I measured with the 8Ω resistive load alone. The three curves on each graph represent (A) the 8Ω resistive load from

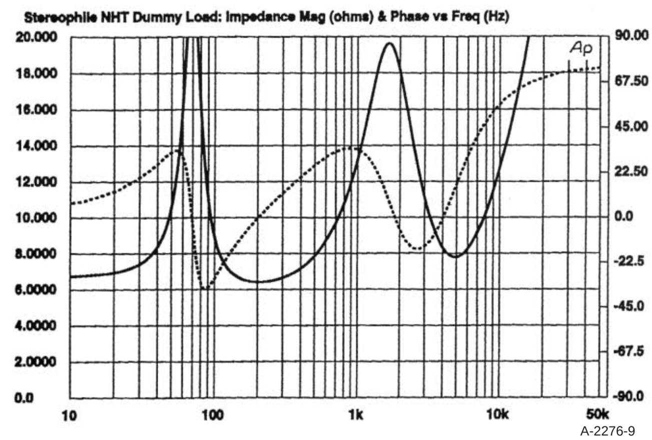


FIGURE 9: Kantor speaker simulator impedance/phase plot.

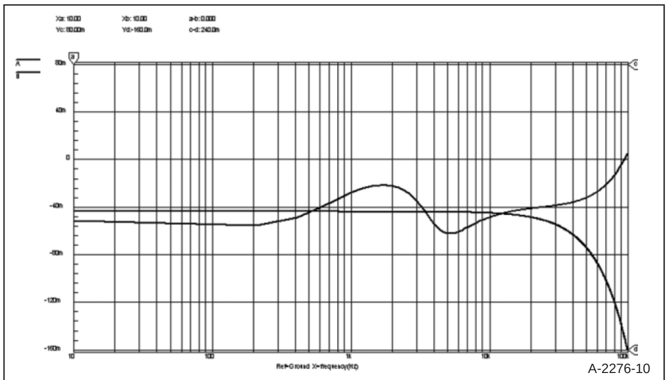


FIGURE 10: Frequency response, 8Ω resistor versus Goertz MI 2 (S-S).

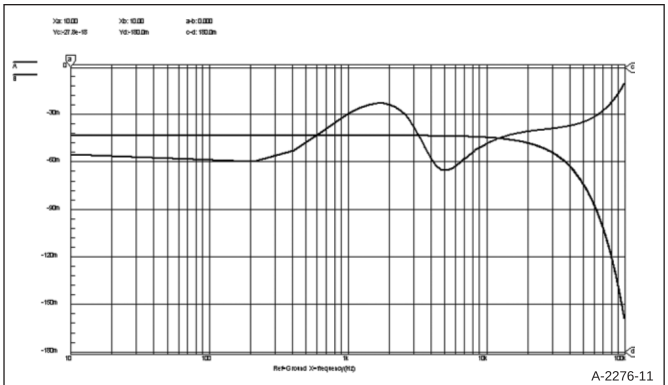


FIGURE 11: Frequency response, 8Ω resistor versus Kimber 8TC (S-S).

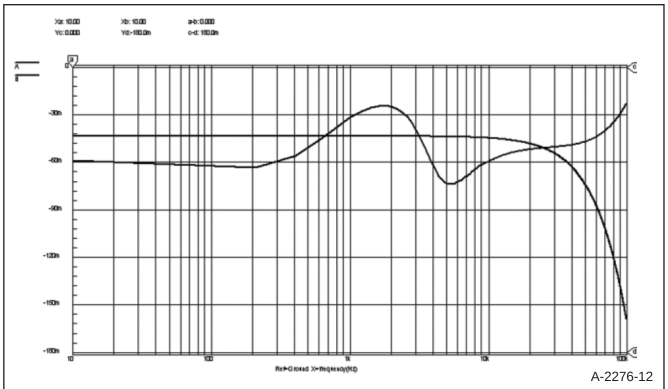


FIGURE 12: Frequency response, 8Ω resistor versus Supra 3.4/S (S-S).

TABLE 1

MEASURED AND CALCULATED PARAMETERS

Cable	Goertz (MI 2)	Kimber (8TC)	Supra (3.4/S)	Monster (12)	Zip (16)	Units
Measured C	2,090.0	836.0	944.0	61.7	66.5	pF
Dissipation Factor	0.002	0.001	0.069	0.056	0.063	%
Measured L	54.0	86.0	694.0	580.0	606.0	nH
Measured R	4.80	21.00	27.00	12.00	23.90	mΩ
Length	3	8	8	3	3	ft
AWG equivalent	10	9	12	12	16	
Insulation	PET	TFE	Stabil.PVC	Linear PE	PVC	
Leakage current	0.3	0.6	1.1	1.1	0.7	pA
Rparallel	38.1	190.5	2.4	46.1	38.0	MΩ
C per ft	696.7	104.5	118.0	20.6	22.2	pF/ft
L per ft	18.00	10.75	86.75	193.3	202.0	nH/ft
R per ft	1.60	2.63	3.38	4.00	7.97	mΩ/ft
Characteristic Zo	5.1	10.1	27.1	97.0	95.5	Ω
Lkg per ft.	0.1	0.1	0.1	0.4	0.2	pA
Resonant Frequency	47	59	20	84	79	MHz

TABLE 2

MANUFACTURER'S SPECIFICATIONS VERSUS MEASUREMENTS

Parameter	Goertz spec	measured	Kimber spec	measured	Supra spec	measured
C per ft	950pF	697pF	100pF	105pF		
L per ft	6.0nH	18nH	50nH	10.8nH	61nH	87nH
R per ft	2.2mΩ	1.60Ω	2.2mΩ	2.63mΩ	1.55mΩ	3.38mΩ
Zo	2.5Ω	5.1Ω				

the initial tests, (B) the THD+N at the amplifier terminals with the cable connected to the speaker, and (C) the THD+N at the speaker terminals.

Figure 2 shows the THD+N versus frequency for the Goertz MI-2, *Fig. 3* the Kimber 8TC, *Fig. 4* the Supra 3.4/S, *Fig. 5* the Monster 1190MC, and *Fig. 6* the 16-gauge zip cord. Bear in mind the varying lengths here.

I intended to measure voltage drop and phase shift with each cable installed in one channel of my audio system. However, with 2V RMS at the amplifier terminals, the voltage drop at the speaker end of the worst-performing cable (8′ zip cord) was 71mV at 20Hz, 76mV at 211Hz, 37mV at 1kHz, 87mV at 5kHz, and 63mV at 20kHz. At 5kHz this is a –0.4dB loss with a phase shift of 0.3°. The attenuation and phase shift in the 3′ lengths of test cable were much smaller. In order to obtain graphical information on comparative 8′ lengths, I decided to model the cables as part of an audio system in SPICE.

SPICE SIMULATIONS

Using the measured parameters, I modeled each cable as part of a SPICE model audio system in order to determine its effect on magnitude and phase. I used both solid-state and tube amp models. The cable models are T-section equivalents based on my actual measurements. The loudspeaker model is the one proposed by Ken Kantor[2] with a nominal impedance of 8Ω.

Caveat: SPICE does not model the back-emf of the driver or the many low-level mechanical resonances present in a real loudspeaker. You should consider these simulations comparative rather than absolute.

Figure 7 shows the solid-state system model, based on an NAD-214 amplifier, for which I have the service manual and thus the schematic. The closed-loop output impedance is 0.04Ω at 1kHz, increasing to 0.16Ω at 20kHz.

Figure 8 shows the tube amp system model, based on the high-frequency model proposed by Scott Reynolds[3]. It has an output impedance of about 0.6Ω at 1kHz. Other tube amp SPICE models are available on the web if you want to experiment[4].

Each amplifier model is set for an open-circuit output voltage of 2V RMS at 1kHz. I ran simulations with an 8Ω resistor directly across the amplifier terminals, and with each cable in series with the loudspeaker model. I used two identical amplifier channels in the simulations to avoid interaction between the 8Ω resistor and the cable/loudspeaker models. The vertical scale is in dB relative to 2V RMS.

The nominal 8Ω speaker impedance has peaks of >20Ω at 70Hz and above 15kHz, 18Ω at 1.5kHz, and reaches down to 6.5Ω at 200Hz. *Figure 9* displays the impedance/phase graph.

Figure 10 shows the modeled response of the Goertz MI 2 as a composite (not bi-wired) 8′ cable. The top line at 100Hz is the response across the 8Ω resistor directly across the amplifier terminals. The output is a flat –0.043dB over the audio band (1.99V at the resistor), until it drops due to the increasing impedance of the amplifier's RLC network outside the feedback loop. This network is designed to make the amplifier inherently stable with all speaker loads.

The MI 2 cable shows the peaks and dips as a result of the complex impedance of the loudspeaker model. These excursions vary from –0.021dB at 1.8kHz to –0.062dB at 5.2kHz, representing voltages of 1.996V to 1.986V, respectively, at the speaker terminals. The phase shift at the loudspeaker terminals over the audio band (not shown)

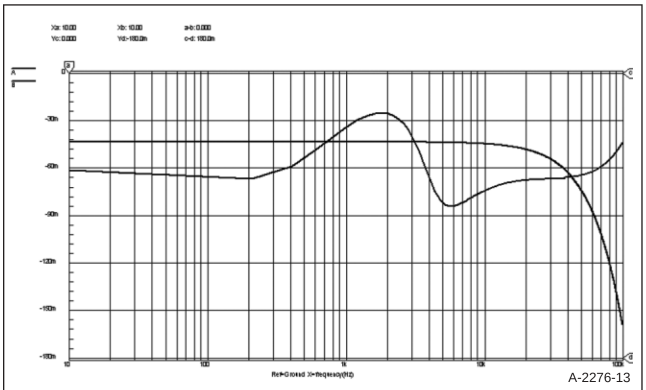


FIGURE 13: Frequency response, 8Ω resistor vs. Monster 1190MC (S-S).

varies from +0.099° at 639Hz to –0.182° at 3.5kHz, using the phase shift across the 8Ω resistor as a 0° reference.

In order to determine phase shift at 20kHz as accurately as possible, I substituted the skin effect resistance for the DC resistance in the cable model. The rise in response above 30kHz is due to the increasing impedance of the loudspeaker, which rises faster than that of the amplifier. Note that if the values of L, R, and C specified by AlphaCore for the M-1 are used in the cable model, the excursions are –0.018dB to –0.059dB with the phase shift varying between +0.092° and –0.180°. This is a negligible difference.

Figure 11 is the response model for the 8′ Kimber 8TC. The peaks and dips as a result of the complex impedance of the loudspeaker model vary from –0.023dB to –0.065dB. The phase shift at the loudspeaker terminals varies from +0.111° to –0.180°.

Figure 12 depicts the response model for the 8′ Supra 3.4/S. The peaks and dips as a result of the complex impedance of the loudspeaker model vary from –0.024dB to –0.073dB. The phase shift at the loudspeaker terminals varies from +0.111° to –0.224°.

The next 8′ modeled cable is the Monster 1190MC 12 gauge cable. *Figure 13* shows the peaks and dips as a result of the complex impedance of the loudspeaker model, which vary from –0.025dB to –0.084dB. The phase shift at the loudspeaker terminals varies from +0.107° to –0.286°.

Finally we come to the 16-gauge zip cord (*Fig. 14*). The peaks and dips as a result of the complex impedance of the loudspeaker model vary from –0.032dB to –0.098dB. The phase shift at the loudspeaker terminals varies from +0.150° to –0.297°.

I took the zip cord and loaded it with the 8Ω resistor rather than the speaker model (*Fig. 15*). The response across the resistor, which was –0.043dB at the amplifier terminals, is –0.056dB at the end of the 8′ zip cord (bottom trace at 2kHz), with a slightly greater –.062° lagging phase shift at 1kHz. Compare this linear response to the rises and dips for that of the cable terminated with the speaker model.

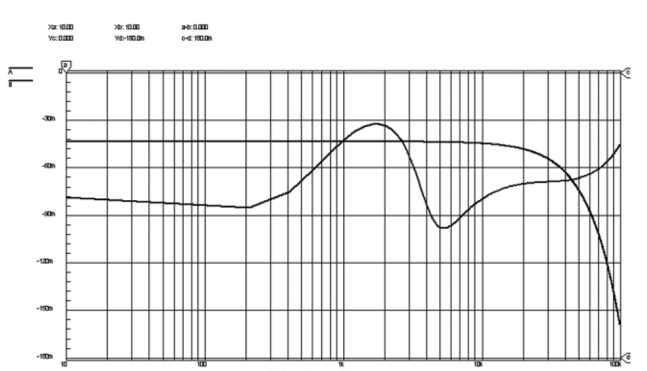
I summarize the SPICE simulation magnitude and phase shift data for the solid-state amplifier models in *Table 3*.

Figure 16 shows the Goertz MI 2 (L, R, and C as measured) feeding the tube amp model, compared with an 8Ω resistor at the output terminals of the tube amp. Again, I used two separate amplifier channels to prevent interaction. You can see the greater degree of interaction between the amplifier and the complex impedance of the loudspeaker. The flat portion of the 8Ω resistor load is down to 1.862V or –0.63dB. The combination of cable and loudspeaker varies from –0.041dB at 34kHz to –0.694dB at 211Hz. The phase shift at the loudspeaker terminals varies from +1.77° at 6.5kHz to –1.04° at 2.8kHz.

The combination of the 8′ Kimber 8TC cable and loudspeaker models (*Fig. 17*) varies from –0.041dB to –0.700dB. The phase shift at the loudspeaker terminals varies from +1.80° to –1.04°.

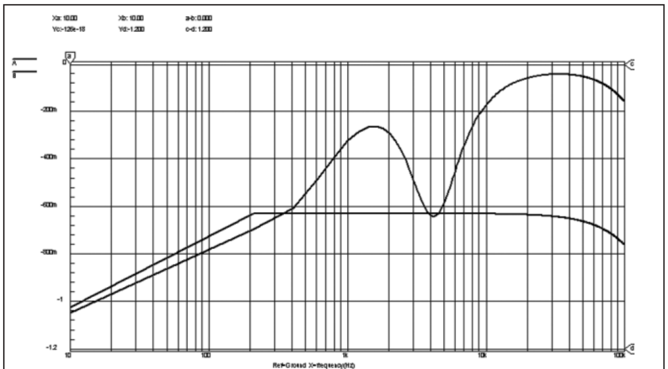
The combination of the 8′ Supra 3.4/S cable and loudspeaker models (*Fig. 18*) varies from 0.052dB to –0.702dB. The phase shift at the loudspeaker terminals varies from +1.76° to –1.06°.

The combination of the 3′ Monster 1190MC cable and loudspeaker (*Fig.*



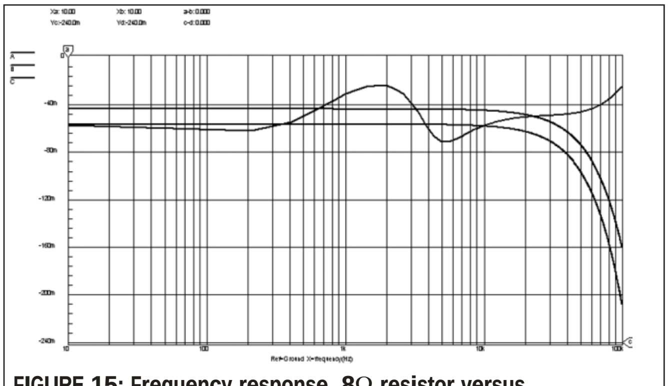
A-2276-14

FIGURE 14: Frequency response, 8Ω resistor versus zip cord 16 (S-S).



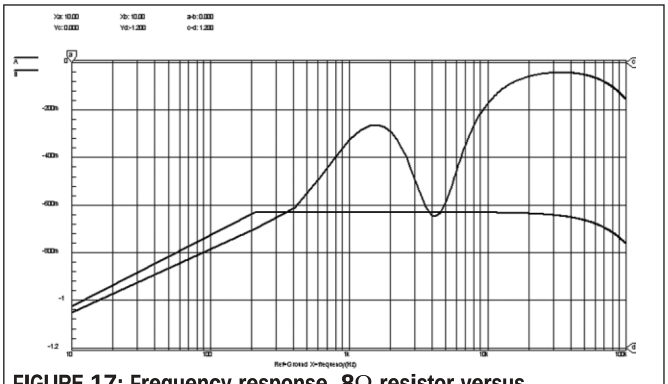
A-2276-16

FIGURE 16: Frequency response, 8Ω resistor versus Goertz MI 2 (tube).



A-2276-15

FIGURE 15: Frequency response, 8Ω resistor versus zip cord 16 with 8Ω load, speaker load (S-S).



A-2276-17

FIGURE 17: Frequency response, 8Ω resistor versus Kimber 8TC (tube).

19) varies from –0.068dB to –0.705dB. The phase shift at the loudspeaker terminals varies from +1.69° to –1.10°.

Using the 16-gauge zip cord in the same tube amp simulation setting (*Fig. 20*) shows the response varying from –0.070dB to –0.720dB. The phase shift varies from +1.73° to –1.12°.

The SPICE simulation magnitude and phase shift data for the tube amplifier models are summarized in *Table 4*.

The increase in skin effect resistance at 20kHz seems to have less consequence than you might imagine. The increase in inductive reactance from 1kHz to 20kHz has a much greater effect on the series impedance than does skin effect. The increase in the RL series impedance component of the six cables from 1kHz to 20kHz is listed in *Table 5*. Note the much higher percent increases for the last three stranded cables, with their higher inductance. The first two cables have a double advantage due to their geometry: low inductance and low skin effect.

CONCLUSIONS . . . SORT OF

Looking at *Figs. 2* through *6*, you can see the distortion levels decrease in every case at the peaks of the loudspeaker impedance when compared with the essentially flat distortion curve for the 8Ω resistor. At the high-frequency end of the curve, where you might expect the rising cable impedance to have an effect, the distortion is essentially identical to that of the resistor alone. The more pronounced rise in distortion at the low frequencies, where the cable is essentially a resistor, may be due to woofer distortion in the satellite speaker, whose –3dB point is 57Hz.

The distortion excursions are lowest with the short 3′ flat wire cable, and highest with the 10′ length of Monster cable and 8′ length of 16-gauge zip cord. In the mid-range the measured distortion is lowest with the 10′ Monster cable than with the other cables, but its low-frequency distortion is also one of the highest. Of the three 8′ cables, the Kimber has the lowest overall distortion excursions. This suggests that short lengths of low resistance, low inductance cable are going to be the most accurate.

There is no question that there are also measurable LRC differences between these three speaker cables. However, the 1kHz impedance of even the 10′ length of Monster cable is less than the 40mΩ source impedance of the solid-state amp and far less than the impedances of the tube amp and

TABLE 3

SPICE SIMULATION RESULTS—SOLID-STATE AMPLIFIER

	Goertz MI 2	Kimber 8TC	Supra 3.4/S	Monster 12	Zip 16	Units
min attenuation	–0.021	–0.023	–0.024	–0.025	–0.032	dB
at Freq =	1.8	1.7	1.7	1.8	1.8	kHz
max attenuation	–0.062	–0.065	–0.073	–0.084	–0.098	dB
at Freq =	5.2	5.1	5.4	5.8	5.3	kHz
max + phase shift	0.099	0.111	0.111	0.107	0.150	degrees
at Freq =	639	623	623	631	631	Hz
max – phase shift	–0.182	–0.180	–0.224	–0.286	–0.297	degrees
at Freq =	3.5	3.3	3.5	3.6	3.5	kHz

TABLE 4

SPICE SIMULATION RESULTS—TUBE AMPLIFIER

Cable:	Goertz MI 2	Kimber 8TC	Supra 3.4/S	Monster 12	Zip16	Units
VT min atten	–0.041	–0.041	–0.052	–0.068	–0.070	dB
at freq =	34	34	34	34	34	kHz
VT max atten	–0.694	–0.700	–0.702	–0.705	–0.720	dB
at freq =	211	211	211	211	211	Hz
VT max + phase shift	1.77	1.80	1.76	1.69	1.73	deg
at Freq =	6.5	6.4	6.5	6.5	6.5	kHz
VT max – phase shift	–1.04	–1.04	–1.06	–1.10	–1.12	deg
at Freq =	2.8	2.8	2.8	2.8	2.8	kHz

loudspeaker, especially the loudspeaker. Only the 16-gauge zip cord impedance is higher than the solid-state amplifier source impedance. At 20kHz, the zip cord and the long Monster cable impedances are both higher than the solid-state amplifier source impedance. In all cases, the variations in speaker impedance across the audio band result in voltages at the speaker terminals that vary above and below those across the fixed 8Ω load alone.

CHARACTERISTIC IMPEDANCE

Any transmission line exhibits a certain amount of loss due to the conductor resistance and dielectric leakage current loss in the insulation. Line loss increases when the line termination is not equal to the line characteristic impedance. At frequencies where the transmission line length is an appreciable fraction of the wavelength, a mismatch in line termination from the line's characteristic impedance will produce reflections from the mismatched termination back to the source of the signal. The phase and amplitude of the reflected energy will vary with the degree of mismatch in impedance. The sum of the voltage vectors flowing toward the load is called the incident wave, and the sum of the voltage vectors flowing back to the source is called the reflected wave.

Note that one important consideration here is the ratio of line length *l* to wavelength λ. For audio signals, even the longest practical speaker cables are an infinitesimal fraction of the signal wavelength. The wavelength of a 20kHz audio signal, even after accounting for the drop in signal velocity propagation factor due to the cable configuration, is over 39 kilometers.

The flat wire cable characteristic impedance comes closest to the nominal loudspeaker impedance. However, irrespective of the extremely small *l*/λ ratio, any attempt to match the characteristic impedance of the speaker cable to the system components is also thwarted by the large excursions in speaker impedance over the audio band.

The flat wire cable and the open weave Kimber cable, with their low series inductance and resistance, have lower losses and phase shift than the stranded cables. This is most evident in the simulations with the solid-state amplifier. The closest comparison of the measurable inductance differences between the various cable configurations is with the two 12-gauge cables (Supra 87nH/ft, and Monster 193nH/ft).

Still, the maximum attenuations for the five 8′ cable models differ by less than 0.036dB, and the difference in total phase shift is less than 0.054°. The higher capacitance of the flat wire cable should not cause instability in a well-designed solid-state amplifier. If it does, the problem lies with the amplifier design, not the cable (Goertz does supply those Match Links).

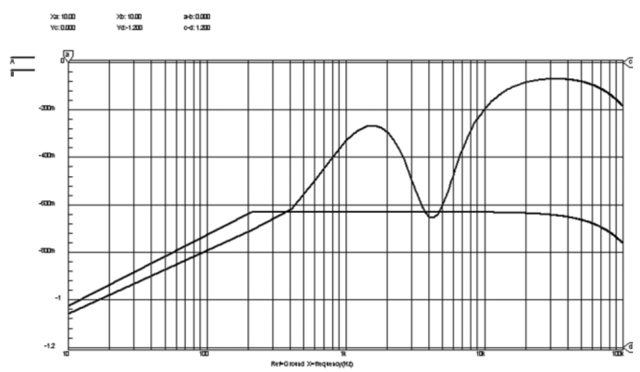
With the tube amp model, the situation is more complicated. Here the interaction of the higher output impedance of the tube amplifier (i.e., lower damping factor) interacts with the complex speaker impedance, producing greater voltage excursions across the loudspeaker than those of the solid-state amplifier. However, the cables just seem to be going along for the ride. There are again negligible differences in the attenuation (0.026dB) or phase shift (0.060°) across the cables, from the amplifier to the speaker terminals, regardless of the cable model used.

Finally, with respect to the differences in insulation materials, there will be a much greater variation in shunt capacitance with the plain PVC zip cord

TABLE 5

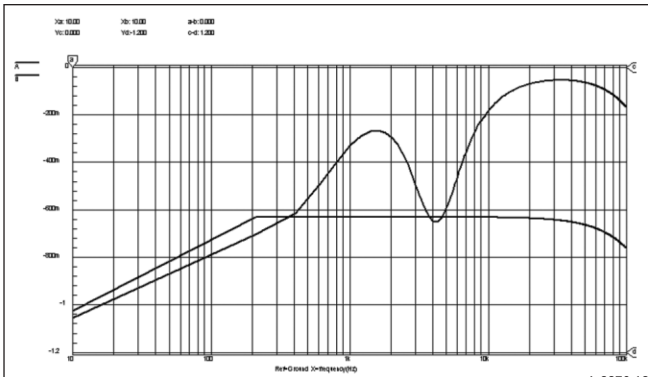
RL SERIES IMPEDANCE VERSUS FREQUENCY

Cable:	Goertz	Kimber	Supra	Monster	Zip 16
RL impedance at 1kHz	12.8mΩ	21.0mΩ	27.3mΩ	33.4mΩ	64.5mΩ
RL impedance at 20kHz	22.7mΩ	27.0mΩ	94.0mΩ	199mΩ	220mΩ



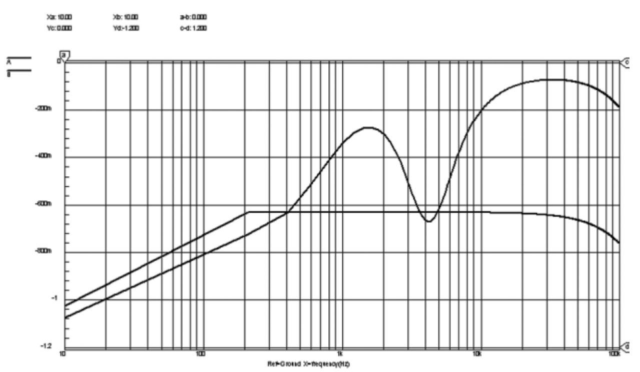
A-2276-19

FIGURE 19: Frequency response, 8Ω resistor versus Monster 1190MC (tube).



A-2276-18

FIGURE 18: Frequency response, 8Ω resistor versus Supra 3.4/S (tube).



A-2276-20

FIGURE 20: Frequency response, 8Ω resistor versus zip cord 16 (tube).

because of its higher dielectric constant and large variation of dielectric constant with frequency and temperature. The highly stable Teflon insulation should give the best electrical performance. All the cables have sufficient insulation thickness to withstand normal handling. ❖

Manufacturer's response:

We have found that it is hard to measure inductance in a short cable due to the inevitability of loops at the hook-up point. Also, because all cable interaction is relative to length, we would have proposed testing a more representative run of 10–15 ft. 3 ½ ft. is a very short speaker cable.

We were happy to see in print that "the capacitance of the flat cable should not cause instability in a well-designed solid-state amp." We share the conclusion that if it does, the problem lies with the amplifier, not the cable. Unfortunately a few designers still violate Nyquist's criteria on allowable amount of negative feedback. A good solid-state amp supply should not oscillate.

Ulrik Poulsen
Alpha-Core, Inc.

First, my compliments on a technically well-carried-out and well-explained test, giving a scientific impression.

Supra Ply is of a sandwich design for low inductance, and we know that even a lower inductance can be achieved with the ribbon design. In fact, we used aluminum tape, flat to flat, for conductors for the measurements of the Ply design concept during the R & D. However, once the theory of low inductance = low transient loss was confirmed, we began working on a user friendly design, which also was a compromise regarding the capacitance not to be too high to risk oscillations in sensitive amplifiers. Furthermore, as we have our own in-house production (different from most produced in China), we made a production friendly solution to ensure value for the money.

The final result was the braided flat conductors of Supra Ply. That was in 1993 when "high-end" speaker cables still were of wide-spaced design. Supra was first with speaker cables in 1976 and first with low-inductance design in 1993.

Tommy Jenving
President
Supra

REFERENCES

1. In order to minimize the skin effect at high frequency, a special stranded wire called Litz wire is employed. Litz wire was designed to minimize the losses in high frequency, high current applications. Litz is an abbreviation for litzendraht, which is German for stranded-wire. Litz wire is composed of separately insulated strands of very fine wire. It is wound so that every strand occupies, to the same extent, every possible position in the cross section of the wire. It is this special construction and the insulated strands that differentiate Litz wire from conventionally stranded wire.

Litz stranding is designed to equalize the flux linkages, and therefore the inductive reactance, of each strand in the wire. This results in the current being distributed evenly among all the strands. Litz wire is typically designed for a specific frequency range. AWG–28 strands may be used for frequencies below 1kHz, while strands as small as AWG–48 are used for frequencies in the megahertz region. The goal of the Litz wire designer is to make the resistance and reactance equal at the design frequency in order to maximize the Q of the cable. Sufficient strands of the selected size are then used to carry the maximum design current.

2. J. Atkinson, "Real-Life Measurements", *Stereophile*, August 1995, Figs. 1, 2. The full text of the original article can be found at www.sterephile.com/showarchives.cgi?60.

3. S. Reynold, "Vacuum-Tube Models for SPICE Simulations," *Glass Audio*, Vol. 5, No. 4, 1993.

4. LTSpice and vacuum tube models, www.duncanamps.com/technical/ltspice.html.

SOURCES

Alpha-Core, Inc.
915 Pembroke Street
Bridgeport, CT 06608
(800) 836-5920, (203) 335-6805
FAX (203) 384-8120
e-mail: sales@alphacore.com
Goertz MI-2 MSRP $21.50/ft pair single, $43.00/ft pair bi-wire.
Terminations: Rhodium spades/banana plugs/pins $48 single, $63 bi-wire. Silver spades $58 single, $85 bi-wire.

Kimber Kable
2752 South 1900 West
Ogden, UT 84401
(801) 625-5530
e-mail: info@kimber.com
Kimber 8TC MSRP $420/pair with angled WBT 0645S banana plugs
Terminations: PostMaster spades, WBT straight or angled bananas

Supra Cables/Jenving Technology AB
Bastebacka 112-113
S-459 91 Ljungskile, Sweden
+46 (0)522-698990
e-mail: supra@jenving.se
Supra Ply 3.4/S $180 2 × 8′ with spades
U.S. distributor: Tonian Labs,
www.tonianlabs.com

Monster Cable Products
455 Valley Drive
Brisbane, CA 94005
(415) 840-2000
e-mail: webmaster@monstercable.com
Monster 1190MC $1.25/ft without terminations

SPEAKER CABLE LISTENING TESTS

By Muse Kastanovich

Cable is the purist-audiophile's favorite tool for manipulating the frequency balance and other qualities of his/her system. Active equalizers/tone controls are just not transparent-sounding enough for many listeners to be considered as part of a high-quality system. What is left to balance the response and sound of the system in the audiophile's listening room is choice of components, especially cables, which are less expensive than components, and so are easier to switch when you need a change. Cables interact with components to at least some degree, so their sound is always context-sensitive, which is why most outlets allow you to audition them with your own equipment.

For this review I first listened to a selection of high-end cables from the US company Kimber and the Supra company out of Sweden. Both are well-established, reputable audio companies. A comparative study lets you test many different qualities in different models, and also allows the ear to hone in on the exact characteristic sound of individual favorites.

LISTENING TEST

I listened to the Supra Ply 3.4/S in comparison with my reference Goertz MI 2 Veracity 3′ biwire pair. The Supra's treble was just a little grainy, and its transients were a touch rolled off. The midrange and lower treble were more prominent, and the bass sounded rich but a little flabby. The soundstage was more vague than that of the Goertz, and reverb was not as pleasantly noticeable. Overall, the Supra was darker and less involving.

After changing back to the Goertz for more reference listening, I then switched to the Kimber 8TC speaker cables. The Kimber's midrange was a bit less meaty, but the bass was more solid. Dynamics sounded equally good, though transients were not quite as coherent. The soundstage was very vivid, with good depth but not quite as much width. The top octave was not as present nor as clean as with the Goertz, but the treble was more prominent.

WRAP-UP

I found the Supra Ply 3.4/S was not the best match for my system. It sounded too veiled and soft for me to recommend it as a competitor to the others. However, it is more affordable than the others and may be a decent choice at entry-level. All of the Supra cables have a pastel "ice blue" color jacket that is absolutely delicious to the eye. It's unfortunate that they were not as delicious to my ears, because I really wanted to like them.

I found the Kimber 8TC, which has been available for many years, to still be a good choice in its price range. Its very dynamic, alive, solid sound made me quite happy when I used it to connect my 804s. It is positioned in a medium price range, and you would be hard-pressed to find a better overall value. I recommend taking a listen to this speaker cable, which should match well with many different systems.

UPGRADE

For about three years, I have been using a bi-wire pair of Goertz MI-2 flat speaker cables, made by Alpha-Core, as the reference wires in my system. So you might imagine I was excited when I heard some months back that they had introduced a new Teflon insulated version called the T-series.

My long-term reference cables are an unterminated (bare-wire ends), 3.5′ long, bi-wire pair of Goertz MI-2 Veracity, insulated with a mix of Teflon and polyester terepthalate. Each pair of flat OFC copper foils is sandwiched together to make a ribbon about ¾″ wide and only about 1/16″ thick, and has the same conductor cross-section as 10 AWG wire. Of course, there are two ribbons per speaker in a bi-wire setup like mine, though only one is needed per speaker in a normal setup.

The new Goertz MI-2 T-series ($136) cables reviewed here are identical to the older version, except that they are 3′ long and have only pure Teflon insulation around them. They are also terminated with what Goertz calls integral spaces, though this is not an important difference because the spades are made from the bare copper foil of the wire, meaning they are actually unterminated like my older pair.

TEST SETUP

My system consisted of a Rotel RDD-980 CD transport and Assemblage DAC-3 D/A processor connected with a Sound & Video Digiflex Plus BNC cable. Analog interconnects were Kimber Hero unbalanced. Amplification was provided by my home-built STAMINA single-ended MOSFET integrated monoblock amps (based on the Pass Zen/Bride of Zen circuits, output impedance 0.8Ω). Speakers were my modified pair of B&W Matrix 804 (nominal impedance 8Ω, minimum 4Ω).

Alpha-Core included a pair of free R/C Match Links to prevent oscillation in certain amplifiers when using their high capacitance cables. I did not need these with my amps because the ultra simple Zen is rock stable into any impedance.

LISTENING IMPRESSIONS

The new Goertz T-series MI-2 wires appear identical to my old favorites, except for the color of the translucent insulating jacket. A change of insulating dielectric material can theoretically change the sound, considering that a good portion of the electromagnetic signal travels on the surface of the wire at audio frequencies. Indeed, I was pleasantly surprised by the improvements that greeted me when switching from my old MI-2 cables.

Listening to "Achtung Baby" by U2, I noted the T-series had a slightly brighter sound, and was a little crisper and clearer sounding. Transients sounded a little faster, and there seemed to be a greater number of different layers of sound apparent. Its soundstage was slightly wider, and images were a bit more vivid. The midrange was just a touch recessed, and the bass was deeper. Overall, I preferred the faster, more dynamic sound of the new T-series.

ASSESSMENT

I very much enjoyed my time with the Goertz MI-2 Veracity T-series flat speaker cables. I found them to be very detailed, transparent, and dynamic sounding. I highly value these main qualities, which make a system sound closer to live music.

I am purchasing the new T-series review samples, which I will use as my new reference. My old reference Goertz wire seemed just too uninvolving after becoming accustomed to what the new ones could do for the music. This is not to say that the "standard version" Goertz wire is no good (it is still being manufactured); on the contrary I still recommend it, but only for systems that are too bright and need a very mellow-sounding wire. For everyone else, the T-series represents a more involving choice that will keep the excitement level high.

RESPONSE:

We are disappointed that Muse Kastanovich didn't find the Supra speaker cable to work as needed in his system. In my experience with B&W speakers, while wonderful sounding, they tend to be rolled off on top and wanting more power than less. The Zen amplifier is a respected amplifier, but along with its minimalistic design, the power, in my experience, is less than desirable for that speaker. If the Alpha-Goertz cables are a little brighter on top, that would explain their more desired results in Mr. Kastanovich's review, andgiven the Supra's flatter response, a less desired result. As with many things in audio, system matching plays a key, and insufficiently acknowledged, role in listening satisfaction.

Tony Minasian
Tonian Labs
U.S. Distributor (Supra)

ROBERT OAKES JORDAN

Robert Oakes Jordan

The audio world missed the passing of one of its pioneers by several years. I only recently found out that Robert Oakes Jordan passed away in July 1998 during heart surgery. It is not too late to mention some of the accomplishments of Robert Oakes Jordan and Associates.

Bob and James C. Cunningham were the directors of Robert Oakes Jordan and Associates. Jim, a former NBC Radio engineer, is still quite active in the audio world. In the early 1950s, Bob and Jim wrote many technical articles on stereophonic recording techniques in *Audio* (Engineering) magazine and other journals. Their book, *The Sound Of High Fidelity* (Windsor Press), still holds up technically and is a great nostalgia trip.

References to Robert Oakes Jordan and Associates are found in the cover notes of many old LP records and prerecorded tapes. In the early 1950s, stereophonic sound was just beginning to become practical. In those days, the audiophile buzzword was "high fidelity." The concepts of direction and motion were new and unknown. Robert Oakes Jordan and Associates produced most of the stereo demonstration records for the then new medium of stereo.

Robert Oakes Jordan and Associates set about recording just about any imaginable sound, building the first stereophonic sound-effects library. These recordings were utilized in stereophonic sound demonstrations for most of the record companies moving into that medium. Their biggest hit was "Sound In The Round," narrated by NBC announcer Tom Mercein.

Sound In The Round was almost universally used to demonstrate both tape-delivered stereo and the later, LP-delivered stereo. As it began with the sounds of a ping-pong game, they probably are responsible for the phrase "ping-pong stereo." Among the companies they produced stereo demos for were Ampex, RCA, Concert-Disc, Allied Radio, and OmegaTape.

Robert Oakes Jordan and Associates also shows up on a number of music record cover notes as consultants for engineering and recording. Bob once told me that in the early 1950s they assisted John Pfeiffer and RCA engineers for recording in Chicago's Orchestra Hall for RCA's first recordings there.

In later years Bob operated a small home-based technical school for young people in his town of Highland Park, Ill. It was sort of a "Mr. Wizard" TV show come home. Young people obtained hands-on technical training in everything from electronics to ceramics.

Mike Stosich
Esoteric Sound/Rek-O-Kut
EsotericTT@aol.com

IONOFANE FIX

Thanks for your very interesting article about the Ionovac in the August issue ("Care and Maintenance of DuKane Ionovac Tweeters," p. 50).

I recently bought an Ionofane 601. It's not working at the moment because of some fault in the power supply that I haven't been able to track down. The electrode is rather worn and blunt—like the one in Photo 9 of your article.

What are the options for electrode replacement for Ionovac owners? Are they available anywhere in the world? Is anyone still making them? Do you know the names of the alloys that were used in the original electrodes?

Jonathan Cook
brahms1888@yahoo.com

Daniel Schoo responds:

I have not seen an Ionofane speaker in person—only photographs of them. According to Roger Russell on his Ionofane history web page, the Ionofane is essentially a copy of the Ionovac, which preceded the Ionofane by several years. I can't guarantee that the DuKane cell will fit the Fane speaker, but the likelihood is very good.

There are a number of things you can check in the power supply if the speaker is nonfunctional. I would replace all of the electrolytic capacitors even if they seem to be good. Measure the supply voltages and compare them to the values on the Ionovac schematic. Try installing a known good tube and also don't overlook the oscillator cover interlock switch as a possible problem.

One option for electrode replacement is to check eBay for old stock DuKane cell kits. They show up infrequently and tend to rocket up in bid in the final seconds of the auction.

There is another option: Due to popular demand, new manufactured Ionovac electrodes are now available at www.ionovac.com. They also have cleaning kits with a detailed videotape tutorial on how to disassemble the speaker safely to clean or replace the cell. If the Ionovac electrode doesn't fit the Ionofane cell, you can return it unused for a refund.

The alloy DuKane used for electrodes was essentially type 416 stainless steel as determined by a metallurgical lab analysis on a DuKane electrode. It is reported that Telefunken used an alloy called Kanthal—which is from Kanthal AB of Sweden, a manufacturer of high-temperature metal alloys for heating elements and furnace parts. Like the term "stainless steel," Kanthal refers to a broad product line of high-temperature alloys, so it is unknown just which one was used as electrode stock.

SERVO SUB

I very much enjoyed Daniel L. Ferguson's article "A Servo Dual Voice Coil Subwoofer" (Nov., p. 18).

Unlike previous similar projects I am aware of, he deals with the mutual induction of the voice-coils.

For example, "Roaring Subwoofer" by Russel Breden (published in *Electronics World*, February 1997), uses two 10″ units; distortion at 15Hz is quoted as 2%, without mentioning the corresponding acoustic power. An interesting feature of Russel's design is that instead of differentiating the coil voltage it integrates the incoming signal, "as there is no difference in system performance, either in amplitude or phase response, between differentiating the pick-up signal or integrating the output." This may be applicable to Daniel's design to avoid saturation problems with op amp E.

The second point concerns the current sensing method. Is it really necessary to go through a true differential scheme (op amp A), which could be avoided by connecting the 0.1Ω resistor at the ground side of the driver?

Finally, it would also be helpful for experimenters to know the gain of the power amplifier.

I hope this project will be continued. I suggest two different directions: an enclosure with a 15″ driver (the Monacor* SPH-390TC seems adequate) and, if the servo can be sufficiently bypassed at higher frequencies, a two-way very compact monitor with an 8″ unit, which will then also handle the medium range.

Sébastien Veyrin-Forrer
Saint Chartres
France

*Editor's note: Monacor products are not, to my knowledge, available in North America.

Daniel L. Ferguson responds:

Unfortunately, I did not have the benefit of reading "Roaring Subwoofer" prior to writing my initial article. I wish I had. Russel's observation about integrating the incoming signal makes perfect sense. The integral of acceleration is, in fact, velocity. If the sensor voice coil output is fed directly to the servo-summing junction, the system becomes a velocity servo instead of an acceleration servo.

I have experimented with this briefly, and, from what I can remember, stability was improved. I used a low-pass filter to flatten the system response, which acts somewhat like an integrator. However, I could never get the system accuracy obtained using the differentiator method, so I abandoned it. It didn't occur to me to integrate the incoming signal, and I will surely run some experiments on this.

The benefit I would expect is the ability to increase amplifier gain (and system accuracy), since the differentiator shifts the feedback signal 90°. This phase shift should be com-

pletely eliminated using Russel's approach.

I chose the differential amp for current sensing because, initially, I wasn't sure which way to connect the woofer. One direction produces positive feedback and uncontrolled oscillation, the other negative feedback and stability. In seconds, I can reverse the polarity on this part of the circuit by switching leads. It would appear that the configuration you propose could be made to work and thereby eliminate an op amp stage. However, one sure benefit of the differential amp is that it eliminates any DC signal components without resorting to capacitors and their inherent phase shifts, which your proposal does not.

I must add at this point that I'm having second thoughts about using the current sensor at all. The phase shift caused by the differentiator is the problem. The current sensing-circuit measures current (i), not the rate of change of current (di/dt). Therefore, this signal must be differentiated to mirror the system model.

In my latest experiments, I can obtain the most accurate low-frequency curve tracing without it. However, I do plan to continue to pursue a way to incorporate this, because it has the potential to provide the ultimate in accuracy and stability. The highest amplifier gain I have achieved to date is 10, and this is without current sensing.

I am planning a future project article with a very high-excursion 12″ driver. It appears to be a good compromise between box size and output. At this point I have not tamed this beast enough to even consider anything in the mid-bass frequency range. Current sensing will have to be perfected to operate in this range. Maybe some day.

Thanks for a very insightful letter. You have given me a number of things to consider for future experimentation.

ACTIVE CROSSOVERS

I have been a long-time subscriber to *Speaker Builder*, *Audio Electronics*, and now *audioXpress* and wish to offer my congratulations on a great magazine. Also, to pass on a comment regarding active crossovers.

I built a three-way system several years ago—satellites with MTM and separate woofer, with monoblock MOSFET amps for each driver and the Butterworth crossover [your own article and design from some years back in *Speaker Builder*]. I neglected to use fuses in the power supplies of the crossovers and had a diode short out, probably due to dirty AC.

The secondary voltage from the crossover power-supply transformer went straight through to the amplifiers and fried the voice coils of each driver in the left channel. Sadly, the 2″ sealed back version of the Philips dome midrange took a beating, and these are now out of production. I suggest that any readers using active crossovers might consider fuses!

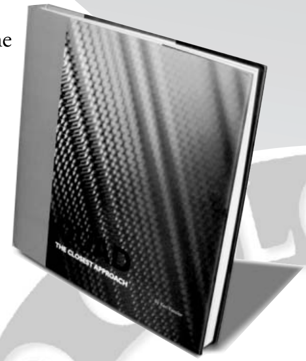
If any readers know of a supplier for those domes, which can be repaired like a tweeter if I can obtain new domes/voice coils, I'd appreciate hearing from them. Philips part # AD0211/Sq. [I think Sq stands for Squawker—British for midrange!]

David Mayfield
maintech@acncanada.net

Quad: The Closest Approach

by Ken Kessler

This much-awaited publication is the life story of Quad, a company started by Peter Walker, who passed away months after the book hit the shelves. Kessler captures the history of Quad through interviews (particularly with Ross Walker and Stan Curtis), reprinted material, and contributions from other industry writers. As a writer for *Hi-Fi News* and *Inside Hi-Fi & A/V*, Kessler takes you through the better part of the 20th Century where the development of high-quality reproduced sound forever changed the audio industry. And although this book contains the history of Quad itself, Peter Walker can now live on through its readers. 2003, 240pp., 12” × 12”, hardbound, ISBN 0-9545742-0-6. Sh. wt: 5 lbs.

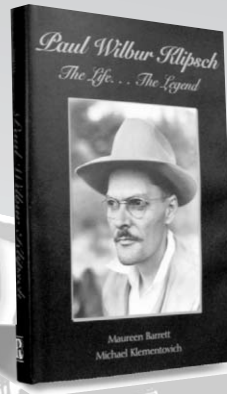


BKIA1 $79.99

Paul Wilbur Klipsch: The Life... The Legend

by Maureen Barrett and Michael Klementovich

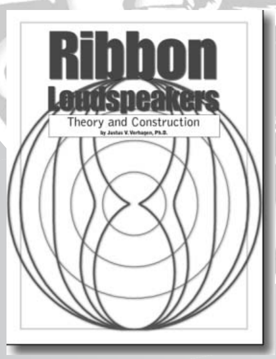
In the only authorized biography of Klipsch to date, Barrett and Klementovich take you through the lifelong journey of Paul Wilbur Klipsch. Ranging from his experience in World War II to the conception and creation of Klipschorns, to being given the Engineering and Science Hall of Fame Award, you are sure to enjoy living Klipsch’s life again in this wonderful rendition. Includes three full photo galleries and diagrams of all twenty-three of Klipsch’s patents. 2003, 208 pp., 6 ¼” × 9 ¼”, hardbound, ISBN 1-58244-226-6. Sh. wt: 2 lbs.



BKPW1 $24.95

Ribbon Loudspeakers

by Justus V. Verhagen



The first book dedicated to these great-sounding speakers. The author favors ribbons, especially for use at mid- and high-frequencies. They are inexpensive and can be easily constructed with a minimum of tools. The book presents the theory and history of ribbon speakers and includes construction details for building your own ribbon loudspeakers. An extensive resource section is included along with a detailed listing of ribbon loudspeaker patents. 2003, 79pp., 8½” × 11”, softbound, ISBN 1-882580-44-3. Sh. wt: 2 lbs.

BKAA65 $24.95

Premium Home Theater

Design & Construction

by Earl Geddes

Describes the ins and outs of designing and building a top notch home theater. Written for those familiar with audio and video, but without formal training, *Premium Home Theater* covers all aspects of the subject including topics like—using a PC as a control center; using a low cost PC projector as the video source; designing a small room with the right acoustics; noise control; and more. Contains al the information you need to assemble an unparalleled home theater experience. 2002, 266pp., 7½” × 9”, softbound, ISBN 0-9722085-1-8. Sh. wt: 2 lbs.

BKGL2 $39.95

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by Ray Alden

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CANTO SIRENA

I enjoyed James Lin's analysis of the differential phono stage in January *audioXpress* ("Canto Sirena, Part 1," p. 32). I built one myself a few years ago, also partly inspired by J.C. Morrison's Siren Song design in *Sound Practices*. I had also read Alan Wright's book.

My design was different in some regards; tubes were 5965 to 6SL7 to 5687 as output. The input stage was cap-coupled to the second stage with low-frequency rolloff below 20Hz. Also, I put the 75µs network first, and the 3180 and 318µs networks were integrated into a direct-coupled cathode-follower output stage formed between the 6SL7 and the 5687.

What I found over time were some odd musical whistlings and phase shifts. These went away when I put a dummy load on the side of the second differential stage that doesn't drive the output tube. This actually balanced that stage and things became more stable and coherent.

James' design may be less troubled by this than mine was due to the less demanding load of the 75µs circuit in this position. Unlike James and J.C., I also made balanced cables from my turntable to my input and therefore had an actual connection to each grid of the input tube from the phono cartridge.

I split the 47kΩ input load into two 23kΩ grid resistors on the 5965 input tube. By the way, the 5965 is an excellent low-noise and low-distortion tube for audio with ultra pure nickel in the cathode for computer applications in days of yore. It is very similar in bias conditions to a 12AT7, but slightly lower mu, and has the same base as the 12AX7, 12AT7, 12AU7, and so on. Good luck to all.

John Day
Austin, Tex.

James Lin responds:

I read Mr. Day's letter with great interest. It sounds as though his design topology differs from mine in that the output stage is a cathode follower rather than a common cathode. In addition, two of the three tubes are different, and the one tube that is common to both—the 6SL7—is used in the second stage instead of the first.

Just from his brief description, there are a number of points of interest. First, the choice of the 5965 input tube. This tube has been used as the third tube in some Conrad-Johnson phono preamps, but is relatively uncommon in audio designs. According to Ludwell Sibley's Tube Lore*, it is a computer-rated version of the 12AV7, although the RCA tube manual specifications for the two are somewhat different. From Mr. Day's description, it sounds as though it deserves a wider audience.*

Compared to the 6SL7, the 5965 has somewhat lower gain (amplification factor of 47 versus 70), but four-fold higher transconductance, which should mean 6dB less noise—an important consideration for a phono input stage. Its plate resistance is also much lower, about the same as the 6SN7 used in my second stage diff amp.

In a split passive RIAA equalization scheme such as used in my design or Mr. Day's, the best equalization stability is achieved by having the higher impedance 318 and 3180µs network following the higher plate resistance tube, and the lower impedance 75µs network following the lower plate resistance tube. In my design the high plate resistance 6SL7 was the input tube, thus the 318 and 3180µs network came first, whereas in his design the lower plate resistance 5965 is the input stage tube, so the 75µs network should be first, as he has done.

A cathode-follower output has the advantage of lower output impedance, which should make his design less sensitive to choice of output cable. The 5687 tube Mr. Day uses is a medium mu dual triode with about the same low distortion as the 6SN7, but a plate resistance less than half of the 6SN7. It has been used in a number of designs from Audio Note, including some of the most expensive amps ever sold.

The trade-off with a cathode-follower output is lower gain. I would guess that Mr. Day's design is closer to 35–40dB at 1kHz versus 49dB for my design, which is still adequate for moving magnet cartridges. On the other hand, one of my goals was a higher gain to give an output closer to that from my other sources. Of course, the cheapest solution would have been to pad down my other sources at the preamp inputs, but what fun is that?

The use of relatively unfamiliar tubes for audio certainly makes things more interesting. Maybe we'll even find that some of them make better music than the 12AX7 and 6DJ8 tubes we are used to hearing, and *wouldn't that be great?*

Incidentally, there is one error in his letter: although I used phono connector inputs, the ground side of the connector is isolated from both chassis and circuit ground—it may not look balanced, but electrically it is. I have balanced cables from my turntable, so the cartridge is directly connected to both grids of the input tube. Morrison shows both balanced and unbalanced connections in his schematic. My schematic shows a balanced connection and the text comments on changes to be made for an unbalanced connection.

In any case, his design sounds very interesting, and I strongly encourage Mr. Day to write it up for publication in audioXpress.

CAR SPEAKERS

I am searching for speakers, but I am not sure about what size I need for my car, a 1997 Saturn, SC2 Coupe. I cannot figure out what size speakers I need. I looked at the speakers, but they have no markings. Can you please help me?

Richard Abston
richard.abston@navy.mil

Crutchfield can help you. . .—ETD

ODE TO A MENTOR

When I moved from New York City down to Florida, I went a year becoming acclimated to a new job and surroundings. I continued reading my hi-fi magazines, but eventually I became lonely for audio DIY friends and longed to meet kindred spirits to talk shop and compare notes with.

It occurred to me that there must be other *Speaker Builder/Audio Amateur* subscribers in my area, so I placed an ad in both magazines inquiring as to whether there were any other like-minded souls out there with whom I could communicate. I received several responses but nothing panned out immediately. I then heard from a fellow by the name of Matt Hamilton. I now consider this to be one of the luckiest days of my life.

As it turned out, Matt is a retired digital electronics engineer who, like me, enjoys tinkering with audio electronics and speakers. And he lives only one mile away from me! We met, became good friends, and have spent many an afternoon together in his garage working on one audio/speaker project or another. Our relationship has been one of teacher/student, with Matt as mentor and myself as protégé for the last ten years.

There is so much that I have learned from Matt over the years, so much that I am grateful to him for. He has always been patient, available, and very giving, willing to share his time with me, always ready to answer another one of my curious questions, always ready to go to the blackboard to draw formulas and diagrams in order to help me better understand a subject.

The topic du jour has not been limited to speakers or audio either: Matt is a walking compendium of wide-ranging knowledge in just about any field you can imagine. Needless to say, whenever I have needed help with anything, from broken doorbells to broken swimming pool pumps and anything else you can dream up, he has always been there to answer my questions and lend a hand and pass his knowledge on to me.

The day I met Matt Hamilton was truly a blessed day. Thanks, Matt, for being my friend.

Angel Luis Rivera
Angel.Rivera@med.va.gov

MULTITONE TESTING

I enjoyed reading Richard Crawford's article entitled "A 1 PPM IM Distortion Analyzer" in the March 2004 issue (p. 8). This is the kind of article I love to see, and believe it is of great value to the DIY community.

It turns out that the three-tone test that Richard presents, using frequencies of 9.6, 10.4, and 20kHz, is actually the Multitone Intermodulation (MIM) test I originated and presented at the AES convention in November 1979, and subsequently published in the JAES in September 1981. My original test employed frequencies of 9.0, 10.05, and 20kHz to produce IM products at 950 and 1050Hz. I'm flattered that Richard independently came up with virtually the same idea 25 years later!

Richard's measurement floor of somewhat less than 0.0001% is very impressive, and bests my floor of about 0.0002%. Of course, my design was somewhat simpler and used older technology. It used three single op amp Wein Bridge sinewave oscillators for the generator. The analyzer used two third-order 2kHz Butterworth low-pass filters and a fourth-order 1kHz bandpass filter for the analyzer. Rejection of the test frequencies was more than adequate, as my measurement floor was also dominated by noise.

My MIM measurements tracked THD–20 quite well for a variety of devices, including a power amplifier, with MIM generally lying about 20dB below THD–20. As Richard pointed out, the MIM test is especially important nowadays with digital equipment, where a test whose stimulus and response are both in-band is a necessity.

Bob Cordell
Holmdel, N.J.

Richard Crawford responds:

My thanks to Mr. Cordell. His original development of the three-tone intermodulation test (MIM) is noteworthy. I recommend the JAES article he mentions, not only for the three-tone test he describes, but also for his discussion of other tests which measure intermodulation. His comparisons of three-tone intermodulation tests and THD-20 (total harmonic distortion at 20kHz) are similar to my own comparisons. A small difference between his instrumentation and mine is that I include a two-tone test to measure third-order nonlinearities.

J-LOW HORN DESIGN

The article by Pass and Kruse on building a single driver loudspeaker is very interesting ("The J-Low Project," Feb. '04 *aX*). However, have they considered transient response effects of a horn that is unloaded on one side? It takes a few cycles for the loading effect of a horn to take effect. In order to keep driver excursion under control for this brief interval, horn designs usually include a small enclosure on the backside of the driver.

Without this loading, the J-Low driver could potentially have extreme transient distortion with cone excursion periodically reaching its physical limits.

Larry Brooks
larrymb@comcast.net

Nelson Pass responds:

The J-Low horn does have a small enclosure on the backside. ❖

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Yard Sale

FOR SALE

Two Orion 225HCCA competition amps/crossover—these car amplifiers are $175 each or $300 for the pair; pair of unused Morel MW164 woofers (double magnets, 75mm Hexatech aluminum voice coils, 4Ω), $100 for pair (originally $80 each). Call Dean at 720-532-2574 or e-mail deanbrenda@comcast.net.

Curcio Audio CD12V D/A Processor (see curcioaudio.com). Complete working unit with upgrades from stock unit. Three Hammond toroid transformers, pre-regulator for HV. Holco resistors, 20 MIT MultiCaps, low ESR caps, asking $1300. Also, power amp project for parts—includes two Hammond 167P60 transformers 60VCT 300VA $100. Contact David at 604 988-6666 or elderton@telus.net.

FRIED C3 two-way TL loaded monitor speaker kit. Two 6.5″ bass-mid drivers, two 0.75″ Hiquphon dome tweeters. Two upgraded series crossovers, binding posts, mounting hardware, construction manual for cabinets. 90dB/1W, 8Ω. $75 + shipping. Contact Dan at 317-299-6024, or fastcat95@juno.com.

Horns: short EV tweet horns from Musicaster PA, approximately 2″ × 6″ mouth, 6.2Ω DCR voice coils. Longer (7″) EV tweet horns, fiberglass oddities, approximately 3″ × 14″ mouth, 7.5Ω DCR voice coils. Make offer. P.S. Parts: 75W transformer; secondaries are 300-0-300V AC, 6.5V AC, with CT, 250V AC, with 247, 243, and 235 V Taps, $15 plus shipping. Small P.S. chokes, ferrite cores: 1.5H, 50mA DC, 35Ω DCR, $5 ppd. 2H, 15mA DC, 60Ω DCR, $5 ppd. CORCOM 6EQ1 EMI filters, 6A at 120V AC, fourth-order balance line filter, $10 ppd. Amps: Altec 1569A mono tube units. Used at V.M.I. for PA. 80W each; rough but complete. $250 each plus shipping. Tom, PO Box 713, Staunton, VA 24402, 540-886-2634.

Two (one pair) Dynaudio 12″ woofers #30W100 in new condition, $200 OBO, e-mail Rick at bjwmathis@comcast.net.

New, top quality parts collection: chips, diodes, transistors, precision resistors, $30; rotary panel mount, autotransformer, 8A, 0-130 VRMS, $28; similar, behemoth autotransformer, 20A, $55; Large two wire/two prong isolation transformer (guestimate 12A), $35. Will consider best offers; e-mail Ken at nad512@erols.com.

Parasound HCA-3500 stereo amplifier 500W/channel, like new condition, $1000 including shipping; Harman Kardon Citation II tube amp 60/60W, extensively modified and upgraded internal and external, documented performance, $500; pair of Hafler DH220 amplifier modules with power supply, no case, ready to use, $120 per set (two available). Call or e-mail for details. Joerg 561-881-8112, jzschirnt@aol.com.

Holman Preamp, $30; DBX 228 NR unit, $20; DBX 119 NR unit, $20; Crown IC-150 preamp, $50; Olson VT FM tuner, $10; Realist VT receiver STA-30, $45; Paco VT integrated amp, $45; Altec 1569A power amp, $100 each; Proton 1100 preamp, $50; Leach double barrel 250W/channel circuit boards and all parts, $125; misc. transf., caps, and so on. Send e-mail for list of other parts, j-rhurley@webtv.net, or shurley@itsmarta.com.

Bozak Symphony, Altec Lansing horns, drivers JBL-L 100s Crown and Janszen Electrostatics, 301-596-5637.

WANTED

Heathkit tt1 tube tester for parts—grid current test switch 6550, EL-37, KT66, KT88, 301-596-5637.

Good used moving coil cartridge (Ortofon, Sumiko, or similar), e-mail type and price to j-rhurley@webtv.net or shurley@itsmarta.com.

Kenwood LO7MII or LO5M output/power transistors: 2SA1007A and 2SC2337A. Purchase or source and substitution information would be welcome. Replacement kit 2SA1494 contents and modification information. Johan Granfeldt, 707-829-7072, e-mail: jgranfeldt@att.net.

JBL L250Ti speakers—mint condition for individual in Australia. Shipping within USA need only be to a forwarding agency warehouse in Los Angeles, e-mail WillnNan@evansville.net.

Tube Sound Equipment: amps, preamps, tuners, crossovers, tubes, tube testers, horn speakers, all working or not, all old/new. Call 850-314- 0321, or e-mail sonnysound@aol.com.

Heathkit stereo power amplifier—good or working condition. Schematic for Precision Fidelity C-8 hybrid preamplifier. George Mueller, mcwill@earthlink.net.

AD INDEX

Audio Aid

BNC Jack Positioner

By Charles Hansen

I make a fair number of projects that use BNC jacks, so I made a tool for aligning the two index pins on the jacks so they are vertical. Not only does this give the finished project a more professional look, it also makes installing and removing test cables easier when the pins are consistently in the same position on a chassis.

Test equipment manufacturers have invested in expensive punches and dies that make chassis holes with flats on one side to make the installation and alignment of jacks, controls, switches, and so on, easy for production assembly. The rest of us must hold the chassis-mounted items in alignment with pliers or wrenches, and hope the components don't rotate out of position as we tighten the retaining nuts. This usually requires rotating the part slightly out of position and allowing for some small amount of final rotation. Unfortunately, this lack of precision can also scratch the finish of the project.

The thin wall of the BNC jack shell presents added problems. You could apply too much pressure while trying to hold it in alignment and squeeze the shell out of round, or damage the two small alignment pins. I tried modifying a standard BNC plug to make an alignment tool, but its slots are designed to rotate for positive engagement, and installing the plug on an uninstalled jack required holding the jack with pliers, so I was back to the original problem.

The BNC jack installation alignment tool (*Photo 1*) works by means of two straight slots that engage the jack alignment pins. This allows you to hold the jack in its final location on the front of the chassis while you tighten the rear retaining nut. It is fast, easy, and produces accurate results.

CONSTRUCTION

Figure 1 shows the drawing for making the tool, with the minimum dimensions needed for a comfortable grip on the tool while tightening the rear nut. I used a ¾″ aluminum bar because it was available. A local machine shop can probably make this tool for under $20, but if you have a drill press and suitable drills, you can make it yourself.

Make sure the aluminum bar is clamped in a perfectly vertical position on the drill press table. Center-punch the aluminum bar and progressively drill the center hole to a final 0.391″ (25/64 drill). Then carefully cut a vertical slot across the diameter of the hole, 3/32″ wide and ⅜″ deep. A standard hacksaw blade makes a slot about 0.05″ wide, so you will need to do some additional work with a small flat file to get to 3/32″ (0.093″). Break all the sharp edges of the slots and the front of the bar with the small flat file. ❖

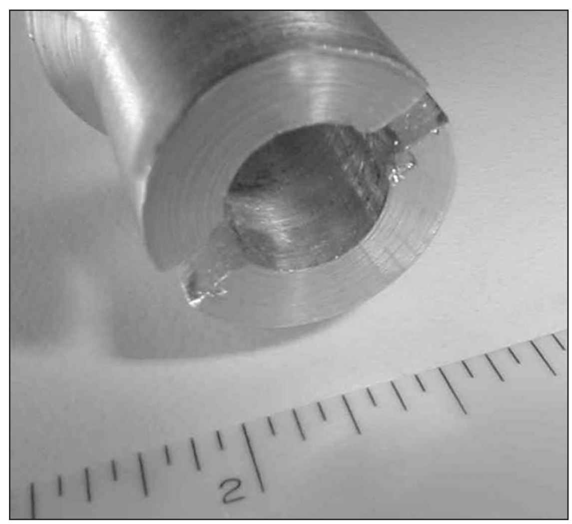

PHOTO 1: BNC alignment tool end view.

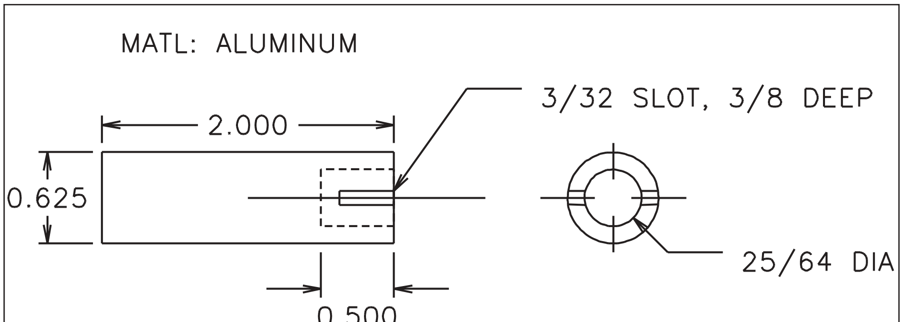

FIGURE 1: BNC alignment tool drawing.

A-2302-1