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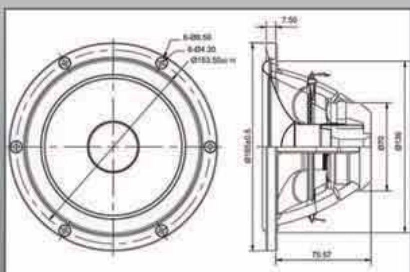

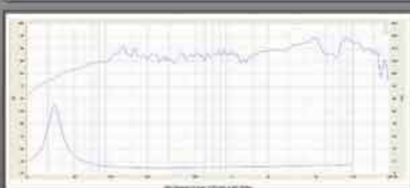

Preliminary Data

Specs :

Nominal Impedance	8 Ω	Free air resonance, Fs	29 Hz
DC resistance, Re	6.2 Ω	Sensitivity (2.83 V/1m)	87.5 dB
Voice coil inductance, Le	0.16 mH	Mechanical Q-factor, Qms	4.9
Effective piston area, Sd	119 cm2	Electrical Q-factor, Qes	0.36
Voice coil diameter	35.5 mm	Total Q-factor, Qts	0.34
Voice coil height	17 mm	Moving mass incl.air, md	13.4 g
Air gap height	5 mm	Force factor, Bl	6.5Tm
Linear coil travel (p-p)	12 mm	Equivalent volume, Vas	44.8 ltr
Magnetic flux density	1.17 T	Compliance, Cms	2.23 mm/N
Magnet weight (NEO)	0.13 kg	Mechanical loss, Rm	0.5 kg/s
Net weight	1.01 kg	Rated power handling	60 watt

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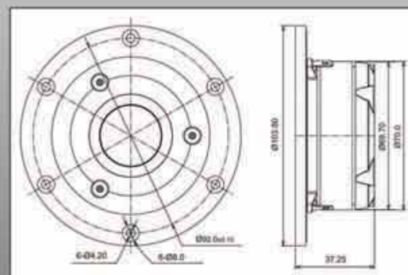

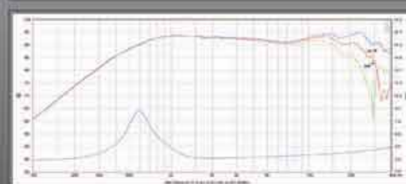

Preliminary Data

Specs :

Nominal Impedance	4 Ω	Free air resonance, Fs	590 Hz
DC resistance, Re	3.0 Ω	Sensitivity (2.83 V/1m)	93 dB
Voice coil inductance, Le	0.02 mH	Mechanical Q-factor, Qms	2.5
Effective piston area, Sd	9.6 m^2	Electrical Q-factor, Qes	1.5
Voice coil diameter	29.0 mm	Total Q-factor, Qts	0.9
Voice coil height	2.0 mm	Force factor, Bl	1.8 Tm
Air gap height	2.5 mm	Rated power handling*	100 Watt
Linear coil travel (p-p)	0.5 mm	Magnetic flux density	1.0 T
Moving mass incl. air, Mms	0.45 g	Magnet weight	0.22 kg
		Net weight	0.5 kg

*(IEC 268-5, Via High Pass Butterworth Filter 2500Hz 12 dB/oct.

We have spent a lot of time trimming the fine details of these magnificent drive units, and we are certain you will achieve

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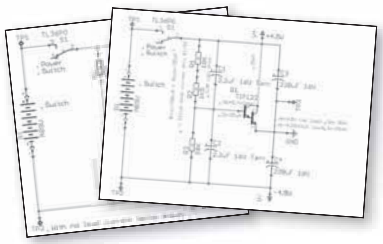

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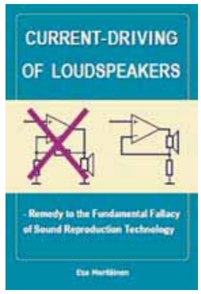

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 By Dick Crawford

The BTO

This version of a Bridged Tee Oscillator (BTO) uses bridged tee tuning, light bulb stabilization, and ICs.

A bridged tee[1] is a circuit which has a minimum transmission at its resonant frequency. A simplified schematic (**Fig. 1**) shows how you can build an oscillator using a bridged tee as the frequency determining circuit. (Ra, Rb, Ca, and Cb) are the bridged tee circuit. Rd is a light bulb used to stabilize the amplitude of oscillation.

The BTO is not new. Peter G. Sulzer used it about 50 years ago, and I built and used a Knight kit BTO. This is a new version of the BTO, using integrated circuits (ICs) instead of tubes, and with improved performance in the audio range.

Table 1 shows the performance of the BTO, which covers the frequency range from 10Hz to 125kHz in ⅓ octave steps. It has an output amplitude of 10V RMS, with a flatness of better than 5% (0.5dB) in the audio range. There is a level control, with additional attenuation (0-80dB) provided by fixed attenuators in 10dB steps.

The total harmonic distortion (THD) of 0.002% at 1kHz is difficult to measure. Most of this distortion seems to be the third harmonic, with the second harmonic and noise about 10dB below the third harmonic. I could not measure any hum. I expect that the THD will be larger at both higher and lower frequencies, but I can only measure THD at 1kHz. Both the short-term amplitude stability and the short-term frequency stability are better than I expected. This

TABLE 1: BTO PERFORMANCE

	Goals	Results
Frequency	10Hz to 100kHz	10Hz to 125kHz
	±3% accuracy	+0.6%, -2%
	1% stability for one hour at 1kHz	0.03%
Amplitude	10V RMS	11V RMS
	1% stability for one hour at 1kHz	0.15%
	0 to 60dB attenuation	0 to 80dB attenuation
Distortion	0.1% THD, 20Hz to 20kHz	0.002% at 1kHz

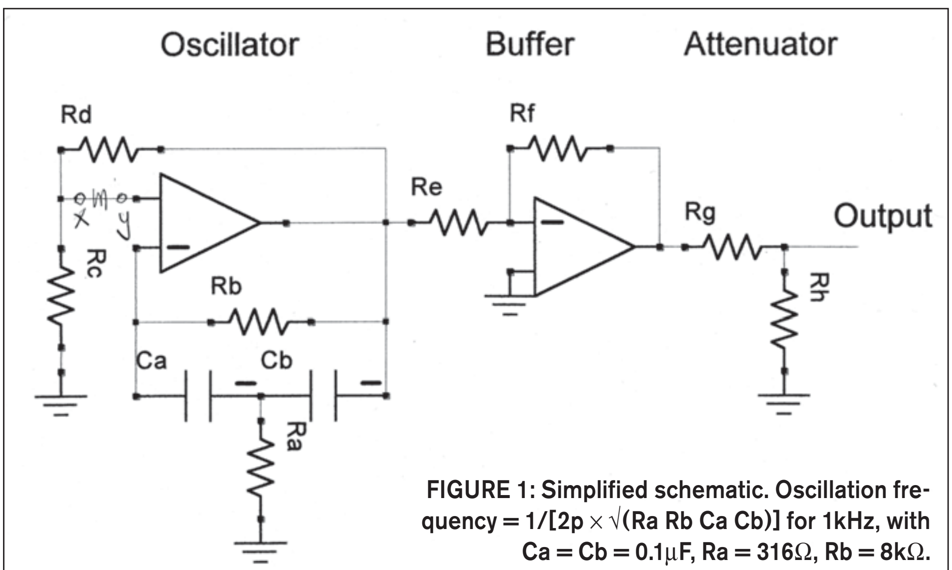


FIGURE 1: Simplified schematic. Oscillation frequency = 1/[2p × √(Ra Rb Ca Cb)] for 1kHz, with Ca = Cb = 0.1μF, Ra = 316Ω, Rb = 8kΩ.

PHOTO 1: The BTO.

seems to be because of the low noise of the OPA2604 op amp.

A single dual op amp (OPA 2604) provides the oscillator and an output buffer. Two more ICs (7820) are used for the regulated ±20V power supplies. A small prototyping printed circuit board contains the ICs, the light bulb, and a few components. Most of the parts are mounted on the switches: the frequency switch (S1), the range switch (S2), and the attenuator switch (S3).

The parts list cost about $70. The prototype (**Photo 1**) shows some of my mistakes. I'm sure you could improve on this.

SIMPLIFIED SCHEMATIC

The heart of the BTO (**Fig. 1**) is an amplifier, in this case an op amp. It should have a high gain bandwidth, low noise, and low distortion. These are desirable characteristics for most audio circuits.

In addition, it needs good output drive capability, because it must drive 10V RMS into both the light bulb and the bridged tee circuit. The light bulb (Rd) is about 1250Ω, and the bridged tee is a reactive load of a couple of thousand ohms, so 30mA peak drive capability is desirable. I chose the OPA 2604 dual op amp.

OSCILLATION

In **Fig. 1**, the points x and y are normally collected, but let's disconnect them for my analysis. If you have an input signal at point y of **Fig. 1**, then the gain of oscillator amplifier at the oscillation frequency is [1 + (Rb/2 Ra)], or 13.5. For oscillation to occur, the signal at point x will need to be slightly larger than the input signal at point y. This will happen if the attenuation of [(Rc + Rd)/Rc] is less than 13.5. Rd is a light bulb whose resistance at room temperature is about 400Ω, and Rc is a resistor of about 100Ω. Thus, at the oscillation frequency, the signal at point x is larger than at point y. If you remove the input signal at point y, and then connect points x and y, you will get vigorous oscillation.

The light bulb (Rd) has a large positive temperature coefficient, which means that its resistance will increase as it heats up. As the amplitude of oscillation increases, the light bulb resistance increases, and the attenuation of the light bulb circuit increases. When the light bulb resistance reaches 12.5 times Rc, the attenuation of (Rd + Rc)/Rc is 13.5, which is the value at which the amplitude of oscillation is stable.

Say the light bulb has a resistance at 1250Ω at an oscillation amplitude of 10V RMS, then Rc will be 100Ω for stable oscillation. Light bulbs are not precision components, so you must adjust the value of Rc to give the desired level of stable oscillation. The light bulb used has a rated life of 3000 hours at 125V RMS, but when operating at 10V RMS its life should be in excess of 100 years.

Let me assure you that this description is greatly simplified. As Bill Hewlett (of Hewlett-Packard Co.) discovered when he invented the light bulb stabilized audio oscillator, the many benefits were balanced by some problems. One well-known problem is "amplitude bounce," in which the amplitude of oscillation "bounces" whenever the frequency is changed. My BTO bounces, but it normally settles in a second or two. At frequencies below 100Hz this settling time may increase to 2 or 3 seconds. Amplitude bounce generally decreases at higher light bulb voltages, which is why my design runs at 10V output.

Another problem with light bulb stabilization is that at low frequencies the resistance of the light bulb will vary slightly during each cycle. This leads to increased THD at low frequencies.

Another characteristic of light bulb amplitude stabilization is that the amplitude will change as the audio oscillator warms up. My BTO has much less heating than tube types, so my design is more stable during startup.

Earlier BTO designs used two gang variable capacitors for tuning, but these aren't readily available today, so I use switched resistors to tune the BTO. For example, when tuning from 1kHz to 12.5kHz, there are 12 sets of resistors arranged in ⅓ octave increments. There are 12 resistors for both Ra and Rb. To switch these resistors requires a two-pole/12-position switch, but this only covers slightly more than one decade, and to cover the full frequency range requires four such decades.

I accomplish this by switching ranges with four sets of capacitors, which vary from 0.001µF at the highest frequency range (10kHz to 125kHz) to 1µF at the lowest frequency range (10Hz to 125Hz). It is important to use high-quality capacitors, those with equivalent series resistance (ESR) of 0.1% or better. Polypropylene capacitors seem to do well, but polyester (Mylar) capacitors are suspect. Using accurate capacitors helps keep the frequency accurate, and the frequency response flat.

The Q of the bridged tee in this BTO is about 2.5. The Q of the more popular Wien network is about 0.3. A higher Q will generally lead to lower THD, but will often result in more

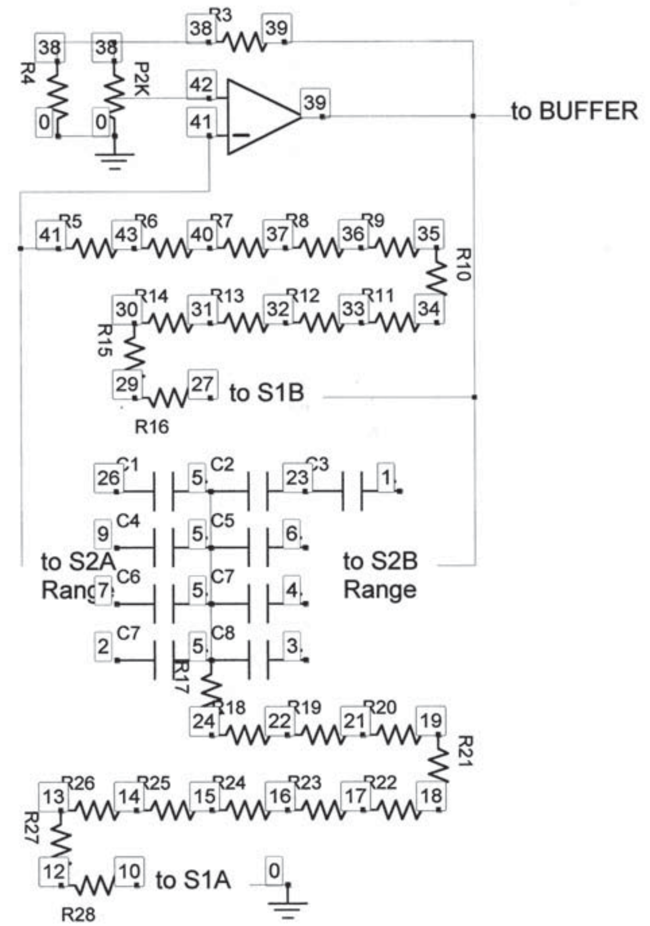


FIGURE 2: Oscillator.

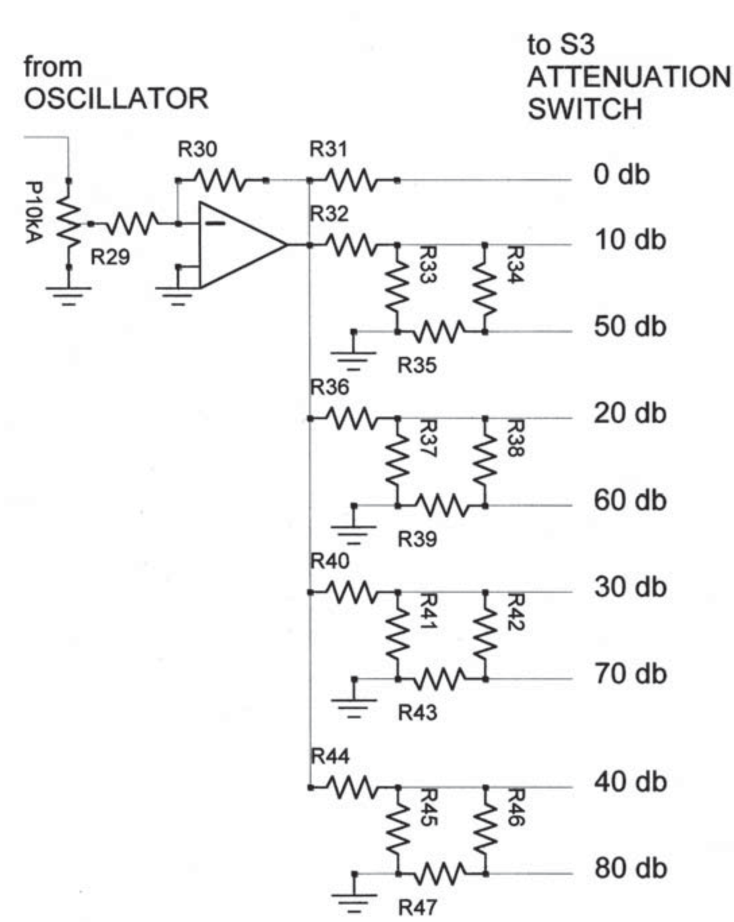


FIGURE 3: Buffer.

amplitude bounce. It may be that a bridged tee with a Q of about 1.5 would be a better compromise for this BTO.

ATTENUATION

Figure 2 is the oscillator schematic. **Figure 3** shows the buffer, which isolates the oscillator from this cruel world. The output level pot is between the oscillator and buffer, which helps keep the output impedance constant. The buffer has an inverting gain of one, achieving low distortion. Following the buffer are the fixed attenuators.

The attenuation varies from 0dB to 80dB in 10dB steps. This gives an output from 10V RMS to 1mV RMS (with output level pot at maximum). This may seem excessive, but I have used 1mV RMS in tests. The fixed attenuators use 1% metal film resistors, but there are often several resistors in series, so the overall accuracy can be as bad as 5% (0.5dB).

As an example of the fixed attenuator, if Rg in **Fig. 1** is 59kΩ, and Rh is 604Ω, then the nominal attenuation is 40dB. S3 selects the desired fixed attenuator. The output impedance of all the fixed attenuators is near 600Ω, which is the output impedance of the BTO.

There is an output at 10V RMS which goes to the sync connector. You can use this output as a synchronizing signal for oscilloscopes and the like.

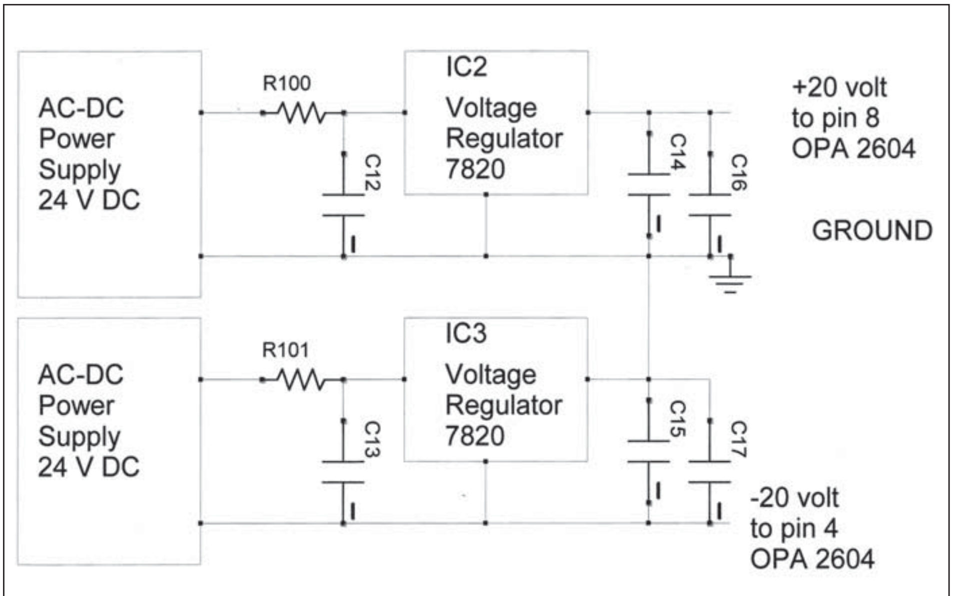


FIGURE 4: BTO power supply.

POWER SUPPLY

The power supply (**Fig. 4**) generates ±20V DC for the OPA2604 IC. This much voltage is needed to generate the 10V RMS for the light bulb. The DC current supplied is less than 25mA. The +20V and −20V power supplies are identical, with the ± voltages being determined by the location of the common (ground) terminal. A pair of 24V AC to DC adapters (wall warts) provide the power to a pair of 7820 voltage regulators. Input RC filters of 62Ω and 1000μF capacitors help reduce power supply noise, as does the 100μF output filter capacitor.

The 1000μF capacitors are mounted (glued) to the metal front panel, while the voltage regulators and the 100μF capacitors are soldered to the same printed circuit board as the OPA 2604 op amp. No heatsinking is required. The 24V AC-DC adapters are plugged into 120V AC outlets, but everything else is mounted to the front panel.

GROUNDS

The sync and output connectors are gold plated, and there is a star ground to the metal front panel at the output connector. The printed circuit board of the BTO (including the power supply) is connected to this star ground, as is the sync ground. The noise measures more than 100dB below the signal at the output connector, and the hum is less. The only improvement I know of is to use a metal enclosure, which might help in the presence of magnetic or RF fields.

Table 2 is the parts list. There are almost 50 resistors (1% metal film), most of which are mounted on either S1 (frequency switch) or S3 (attenuator switch). Most capacitors are mounted on S2 (range switch). There is one capacitor (C3 = 0.018μF) that helps keep the amplitude and frequency correct on the highest frequency range. There are two 220pF NPO ceramic capacitors (C10, C11), which keep the frequency response flatter at high attenuations.

CONSTRUCTION

The circuit itself is simple enough that you should be able to build it, but the switches S1 and S3 are tedious and provide room for error. I haven't designed a printed circuit board for the BTO, but if enough requests come in I will do that, along with more construction details. *aX*

REFERENCE

1. Reference Data for Radio Engineers, 6th Ed., p. 16-11, Fig. 17.

TABLE 2: BTO PARTS LIST

2 each AC to DC adapter (power supply, 24V DC at 100mA, Xicon, 2 each Mouser 412-124013 $6.90

Component	Description	Source	Quantity/ Price
CAPACITORS			
C1, C2	0.001μF/50V, 1% polypropylene	Digi-key P3824-ND	$.63
C3	0.018μF/50V, 2% polypropylene	Digi-key P3915-ND	$.41
C4, C5	0.01μF/50V, 2% polypropylene	Digi-key P3913-ND	$.41
C6, C7	0.1μF/50V, 1% polypropylene	Digi-key P3872-ND	$.15
C8, C9	1μF/63V, 2% polypropylene	Digi-key BC2076-ND	$1.48
C10, C11	220pF/50V, 5% NPO ceramic	Mouser 140-CC50N5-221J-RC	$.15
C12, C13	1000μF/50V, Nichicon UHE	Mouser #647-UHE1H102MHD3	$.98
C14, C15	100μF/50V, Nichicon UHE	Mouser #647=UHE1H101MPD	$.28
C16, C17	0.10μF/100V, 10% ceramic	Mouser 80-C320C104K1R	$.12
INTEGRATED CIRCUITS			
IC1	OPA2604AP dual FET input, 20MHz GB	Digi-key OPA2604AP-ND	$4.28
IC2, IC3	7820FA, +20V regulator, TO200, JRC, Insulated	Mouser #513-NFM7820FA	$.80
POTENTIOMETERS			
P2k	3/8″, 19 turn, 2K	Vishay/Sfernice 72-T93YB-2K	$1.32
P10ka	Alpha 24mm, 10k, Audio taper	Mouser # 31VJ401-F	$1.40
RESISTORS			
All 1% metal film, ¼W, unless otherwise noted.			
Part numbers for Xicon (Mouser) are 271-VALUE-RC, as illustrated by (R4, R23)			
R4, R23	158Ω, 271-158-RC	$.09 each	
R5	6.34kΩ		
R6	1.54kΩ		
R7	2kΩ		
R8	2.67kΩ		
R9	3.3kΩ		
R10	3.92kΩ		
R11	5.11kΩ		
R12	6.81kΩ		
R13	7.87kΩ		
R14	10kΩ		
R15	14kΩ		
R16	15.4kΩ		
R17	255Ω		
R18	62Ω		
R19	80.6Ω		
R20	107Ω		
R21	133Ω		
R22	158Ω		
R23	205Ω		
R24	274Ω		
R25	316Ω		
R26	402Ω		
R27	565Ω		
R28	620Ω		
R29, R30	10kΩ	Mouser # 271-10K-RC	
R31, R35, R39, R43, R47, R37, R41, R45	604Ω		
R32	1.91kΩ		
R33	887Ω		
R34, R38, R42, R44, R46	59kΩ		
R36	5.36Ω		
R40	18.2kΩ		
SWITCHES			
S1	2-pole, 12 positions per pole	Mouser 10WR212	
S1	1-pole, 12 positions per deck, 2 decks	Mouser #105-SR2921F-12S	$4.40
above is Alpha #SR2921F-0112-19R0B-E9-S-B			
S2	3-pole, 4 position, shorting	Mouser 105-2511F-34S	$2.77
S3	1-pole, 12 position, shorting	Mouser 105-2511F-12S	$2.77
MISCELLANEOUS			
Connectors RCA, gold plated, insulated Parts Express 091-1120 2 at $6.89			
Enclosure	8″ W × 6″ H × 3″ D, plastic with metal panel	Radio Shack 270-18/09	$5.99
Knobs	25/32″ diameter, Davies	Mouser #5164-1100	3 at $0.86
Light bulb	R3 GE night light 4, 4W, 3000 hour, C7 clear bulb		2 at $2.99
Printed circuit board		Radio Shack	$1.99
Socket	8-pin DIP	Mouser 575-193308	$.58
NOT LISTED			
fender washers, grommet, feet			

SOURCES
www.Mouser.com
www.DigiKey.com
www.radioshack.com

The Fire Stick Speakers

These easy-to-build speakers include small drivers from Parts Express and Audience.

My early January mailbox offerings included a Parts Express sales flyer, which is always a treat for those of us looking for bargain project drivers. One that caught my attention was an unusual JVC 1″ × 4″ driver claiming to have a full-range response. It also featured two voice coils driving a single cone with a neodymium magnet. This seemed an intriguing transducer, so I ordered ten of them (www.parts-express.com).

Upon their arrival, I unpacked the drivers and found they were housed in a plastic baffle that indicated they were intended for use in flat screen TVs. I attached a couple to the output of my test receiver in the shop and was surprised at the fidelity they produced. Not much low-end, but the midrange and treble were tolerable and you could crank up the volume on them without too much distortion.

I removed them from their plastic housings and examined their construction. They consist of "racetrack" style flat cones with bumps molded into them to provide sound radiation without breakup (**Photo 1**). The rear revealed a bar-style magnet and two voice coils attached to the cone. While they looked fragile, they turned out to be well-made and were capable of taking the abuse they received during the project construction.

"Now, what can I do with these things?" I thought. I laid them out on the workbench and my first thought was "line array." Why not? They don't sound bad singly, and a small array of them could easily be assembled, but they would require some low-end reinforcement.

It just happened that a few days later two of the new model A3 3″ Audience woofers arrived at my door. These little jewels are capable of a flat response from 40 to 20k, and while intended for use in the high-end line array speakers made by Audience, they can be used as standalone full-range drivers or woofers, due to their excellent low-end response (www.audience.com). They are equipped with dual 8Ω voice coils, so they can be wired in any impedance combination desired. The specs of the two drivers were matched well, with their SPL at 80 and 81, respectively. The A3 has an Fs of 70Hz and the JVCs are at 147Hz, so I decided to cross them over at 1000Hz with a simple 6dB first order design.

For this project I did not want a wood enclosure. A few years ago I built some

PHOTO 1: The A3 woofer (top) with the 1″ × 4″ JVC drivers (center). The template (bottom) is used to draw the speaker cutouts.

PHOTO 2: The completed PVC speakers.

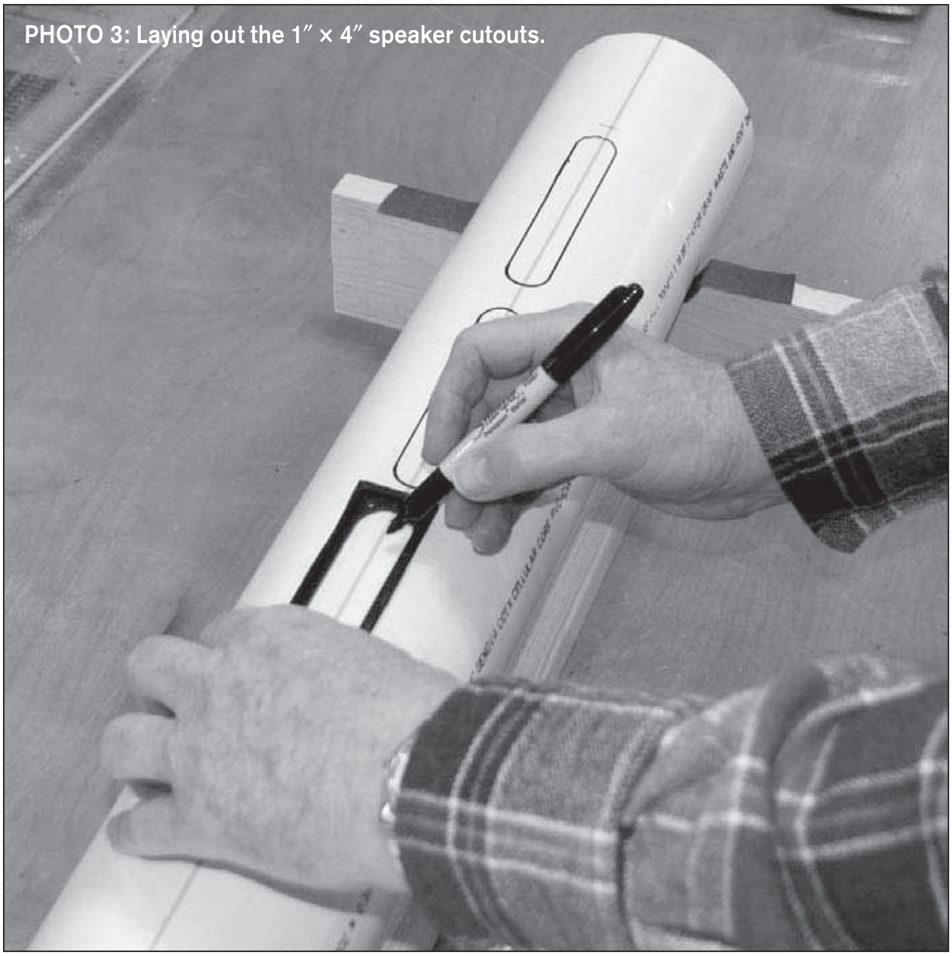


PHOTO 3: Laying out the 1″ × 4″ speaker cutouts.

transmission line speakers using PVC sewer pipe and elected to look at a design using the same material. I came up with a 502 in^3 sealed enclosure using a 40″ piece of 4″ ID Schedule 40 PVC. The walls of this pipe are very dense, so vibration issues would be at a minimum. A 10′ piece of 4″ ID PVC cost $15; add the 4″ caps for $10 and, except for a wood base, you have your enclosure (**Photo 2**).

PREPARING THE PIPE

The first step is to cut two 40″ lengths from the 10′ pipe. You can easily cut PVC with a hacksaw; just be sure to mark it squarely all the way around. I used a piece of masking tape to make the cut mark and rotated the pipe as I made short cuts until it was severed. Use a file or wood rasp to trim the cut ends.

Remove the JVC drivers from their housings, saving the original mounting screws, because you can use them to attach the drivers to the PVC. To facilitate the holecutting for the odd-shaped drivers, I made a template by cutting up and removing a front panel from one of the original housings (**Photo 1**).

PVC pipe has a handy stripe running lengthwise, so use this as the centering guide for the template. Measure 3½″ down from the end and place the top of the template on this mark. Trace the inside outline with a felt-tipped pen. The holes need to be spaced ¾″ apart (**Photo 3**).

Using a high-speed side cutting drill bit, carefully make guide holes along the marked line and run the bit sideways, staying within the line. Use a paper air filter or respirator to protect against some nasty PVC dust. If you can do it outdoors in a breeze, all the better.

After making the first holes, test-fit a driver, and make corrections using a file or rasp. Lay the speaker in the holes as a test and space the template below it to the desired spacing and trace another pattern. Repeat this procedure until you have cut the four holes (**Photo 4**).

You will note that the driver sides don't quite fully touch the enclosure walls. After all, you are laying a flat piece on a curved surface. You can remedy this by using silicone caulk later in the procedure.

Mount the woofer on a baffle board inset into the pipe. The inset is located 3.5″ below the bottom of the last 1″ × 4″ driver. The baffle is a 4¼″ × 5½″ piece of pine or hardwood. The square hole in the board for the A3 driver is 3″ × 2¾″. I cut the driver holes first in the wood with a jigsaw and then cut out the baffle, rounding the edges with a router. Carefully cut the pipe inset to the same dimensions so that the ¾″ thickness of the baffle is even with the surface of the PVC. Cut the cross-section with a hacksaw and the vertical section with the same tool used to cut the 1″ × 4″ driver holes. Smooth the edges square with a file.

The wood baffle board is held in place by two 6¼″ carriage bolts that come through the back of the pipe. Countersink the bolt heads, which are held in place by nuts on the board rear. Fill the bolt head holes with wood putty and sand flush. Paint the board with primer, sand, and then apply the desired final coat. Set them aside to dry. Once all the holes in the PVC tube are cut, test-fit all the drivers and the wood baffle in the openings, and make any necessary adjustments.

The base can be round or square. I happened to have some 7″ diameter wood blanks from a past project and they already had a center hole that my Jasper circle jig and router could use. I routered a ⅜″ groove 4½″ in diameter in the base blank and rounded the edges with the router. The PVC pipe fits nicely into the groove and it makes a sturdy base. You can increase the diameter of the base if you wish, but I found the speakers were not top heavy and 7″ worked fine. Do not permanently attach the base until the wiring is complete (**Photo 5**).

PAINTING THE ENCLOSURES

Finishing the enclosures involved spraying the two pipes, the caps, and the wood base with gray auto primer, and then wiping them down with 000 steel wool. Then I applied two coats of the desired color. I went for a black and red scheme—fire plug red with black tops and base—but you can choose any combination. Let the primer and paint dry at least 24 hours between coats.

With the paint dry, it's time to install the drivers. To protect the paint, I made a simple cradle out of scrap wood and lined the cradle edges with soft cloth. The enclosures can lay in this cradle while you install and wire the drivers.

I first installed the rear speaker terminals, located 5″ up from the base. I used chassis mount banana jack binding posts (Parts Express 091-1140). To mount the 1″ × 4″ drivers, I used the same screws that held them in their original enclosures. I attached the wires starting with the top driver. They are wired in series-parallel to achieve total 8Ω impedance. Lay the driver into the hole and drill a 1⁄16″ pilot hole using the driver mounting holes as a template. Install the same screws that were in the original housings, but do not overtighten them, just snug up the screws.

WOOFER BAFFLE BOARD

Test-mount the baffle board into the inset and mark the boltholes on the inside rear of the pipe. Drill ¼″ holes and attach the board using both a fiber and steel washer on the bolts. Once installed, cut off the excess bolts with a hacksaw. You can also replace the nuts with chrome cap nuts if you desire.

CROSSOVER MOUNTING

I attached the crossover cap and coil to the side walls of the pipe with a large gob of hot melt glue, soldered the feed wires in line, and covered the joints in heat-

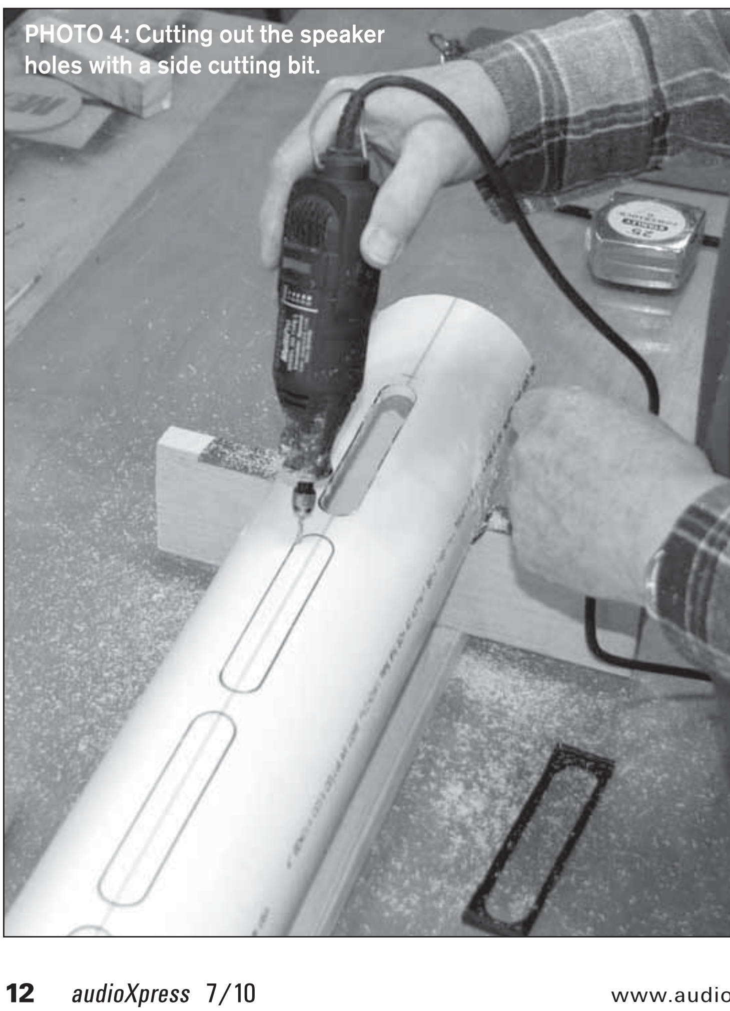


PHOTO 4: Cutting out the speaker holes with a side cutting bit.


PHOTO 5: PVC pipe enclosures ready for painting.

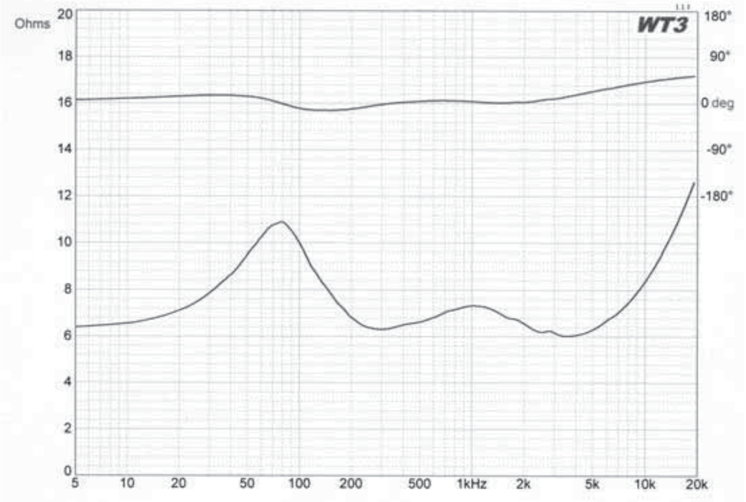


FIGURE 1: Impedance curve of one enclosure.

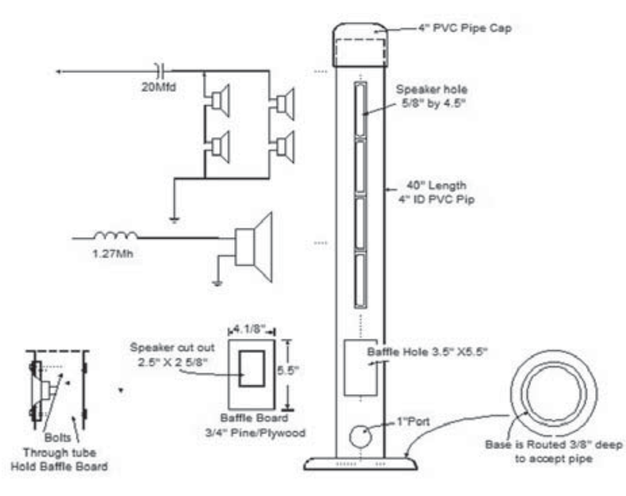


FIGURE 2: System diagram.

shrink tubing. I soldered the woofer and mid/treble leads to the terminals through the bottom. Do not mount the woofer until the stuffing is completed. You can now attach the base to the pipe using a caulking gun and the black silicone caulking described later. You don't need more than a thin bead to get a solid bond.

STUFFING

The stuffing consisted of 8 ounces of polyester added from the top down to the baffle bolts. I used a dowel rod to distribute it evenly. The baffle bolts will keep it from settling and you will need to add more below the bolts in the base and behind the woofer.

Once the stuffing is complete, wire and mount the woofer to the wood baffle plate using small wood screws. Predrill holes for the screws and use the supplied rubber gasket for the woofer.

SEALING THE SPEAKER

Install the pipe cap and push it all the way down. I did not glue it on because once you do, it is on forever. I found that it was snug enough and had no detectable air leaks.

The 1″ × 4″ drivers and the woofer baffle board do not seat solidly to the pipe, so you need to apply caulking. The product I used is a black silicone caulk called *White Lightning Ultra 3000 Silicone, Black*, which is available at most hardware or big box home centers.

Start by cutting off a small part of the caulk spout so that you can eject a narrow bead of the material. Start with the top driver and apply the caulk along the sides first. Keep a roll of paper towels and water handy to clean up around the edges.

Once you've caulked the sides, apply caulk to the tops and bottoms of the drivers, filling the space between them. Rotate the pipe and inspect the joints between the drivers and the pipe; you should not see any air gaps. Do the same around the edges of the wood baffle board. Wipe off any excess with a damp paper towel and inspect for air leaks.

TESTING

Give the caulk at least 24 hours to cure before testing the speakers. I used the WT3 Woofer Tester to check the impedance (**Fig. 1**). If you have a severe air leak, you will note a double peak in the response curve. One way to look for a leak is to play a 30 or 40Hz signal through the speaker while running a wet finger along the edge of the joints. The leak will be revealed by a cool puff of air, and a reapplication of caulk will usually seal it up.

I no longer attempt formal frequency response tests on my projects. I've tried the RTA Computer program, but without an anechoic chamber, the results you get are not of much value due to room reflections or background noise. I have better luck just doing listening tests or using a handheld sound meter and a test CD or signal generator.

The audio from these speakers is fine. The treble is adequate and not harsh. Cymbals and triangles come through clearly in both the near and far field. Their TV sound heritage accounts for their vocal clarity and the A3 does a solid job with the bass without any single note boom. They are ideal for a bedroom or den system, and teenagers would like them for their computer gaming or MP3 playback. *aX*

PARTS LIST

1—10′ piece of PVC schedule 40 sewer pipe
2—4″ PVC pipe caps
1—1″ × 8″ × 36″ pine or hardwood board
4—6″ × ¼″ carriage bolts with four nuts each and fiber and steel washers
2—tubes of black silicone caulk (see text)
8—1″ × 4″ drivers, Parts Express, 269-484
2—Audience Model A3 3″ Woofer, Parts Express, 269-220
1—20mF at 100V non-polarized capacitor, Parts Express, 027-348
1—1.27mH coil, Parts Express, 255-450
2—chassis mount banana jacks, Parts Express, 091-1140

By Bard Kallestad

Designing For Everyone

A call for adherence to standard circuit-design practices.

By having enjoyed many articles in *audioXpress* written by some of the best design wizards of our time, such as Walt Jung, Siegfried Linkwitz, and many others, I was surprised to see an article published in April '09 about a junk box circuit whose reproducibility was questionable. This article ("Keeping the Hobby Alive," p. 38) was fun to read and cleverly written, but it missed a very important electronic principle about providing an artificial ground for a bipolar power supply. That is, the centered ground must be capable of both sinking and sourcing current.

I liked the author's light-hearted, hobbyist approach to a technical problem, that is, providing a decent, battery-powered amplifier to drive quality headsets being fed from a low-cost MP3 player. He used a single, 9V battery with an emitter-follower circuit to provide a center-point ground, making a bipolar supply (almost) to power a simple op amp stage.

However, when a circuit design is published, without any caveats, it is assumed that others will try to copy it as is, and expect it to work. The published design should be workable for everyone with off-the-shelf parts. Good design standards and practices dictate using worst-case specs (specifications) for all of our parts. We need to differentiate between the *science* of electronic design and the *art* of hobby electronics.

I do both and appreciate the finer points of each. For a mass audience, the circuit in question should be capable of working for everyone, not just those lucky enough to have purchased parts that were in the upper 50% of the quality check or are savvy enough to "tweak" the circuit into a useful, working model. At the bare minimum, you should be using worst-case specs for your components.

"WORST-CASE" SPECIFICATIONS

In plain English, worst-case specs are the opposite of the desirable characteristics that you would like to see in your parts. Most spec sheets include minimum, maximum, and typical (nominal) ratings for the various parameters of a device. It is not always the minimum or maximum rating, but the opposite of which rating would be best. When designing, you must use the worst-case parameter, or the most undesirable rating for each parameter.

For example, for transistors this would be the lowest or minimum gain, Beta or h_{FE} spec. For ICs the "worst-case" list would include the highest noise spec, the lowest gain spec, the lowest output current spec, the lowest input impedance spec, the highest output impedance spec, the slowest slew-rate spec, the smallest bandwidth spec, and so on. Consider each part in your circuit to be the worst of the lot.

Design around these parameters for each part and your design will be workable for all of the parts distributed around the world. Of course, be mindful of any adverse effects that the better parts might have in your design. See **Table 1** for examples of some worst-case IC specifications.

TOLERANCES

Common resistors come in 0.1% to 20% tolerance ranges, and you should consider what the effects are of being this far off target, especially in time, frequency, and phase-sensitive areas of your circuits. Also, different material compositions can affect sonic or noise characteristics in circuits.

Capacitors, likewise, have tolerance ranges, and some of them for electrolytic types can be as much as +50%. Additionally, capacitors have other characteristics that you should consider, such as polarity, ESR, and temperature stability specs.

Diodes have current and voltage ranges (forward and reverse), and you should always use a safety margin. Most designers use anywhere from 50% to 200%. Finally, all of your components have temperature characteristics and specifications. You always need to decide on the priority of the different component effects on our circuit design.

I think in terms of macro-factor, mini-factor, or micro-factor. If any circuit part specification and its tolerance represent a major concern, then it becomes a macro factor, which must be addressed and eliminated. Mini-factors have subtle effects and can be improved upon. You will need to evaluate these on a case-by-case situation. Micro-factors are those component effects that are analogous to picking the fly-poop out of the pepper shaker! Perfection can be an endless quest.

TABLE 1: IC specification/characteristics chart.

Symbol	Parameter	Conditions	Min	Typ	Max	Units
Z_{IN}	Input Resistance			150		kΩ
I_{BIAS}	Bias Current	Inputs Floating		100		nA
A_V	Gain		40	50	60	V/V
P_{OUT}	Output Power	THD = 10%, R_L = 8Ω	5	5.5		W
I_Q	Quiescent Supply Current			8.5	25	mA
$V_{OUT\,Q}$	Quiescent Output Voltage			11		V
BW	Bandwidth	P_{OUT} = 2W, R_L = 8Ω		450		kHz
V^+	Supply Voltage		12		26	V
I_{SC}	Short Circuit Current (Note 6)			1.3		A
$PSRR_{RTO}$	Power Supply Rejection Ratio (Note 3))			31		dB
THD	Total Harmonic Distortion	P_{OUT} = 4W, R_L = 8Ω		0.25	1.0	%

The amount of correction required is directly proportional to the magnitude of the negative effect of not correcting it and how this would impact your circuit design. Use common sense and worst-case specifications.

Keeping this concept in mind, refer to the circuit in **Fig. 1** from a recently published article. Drawings are made with EAGLE electronic CAD software by CADSOFT.

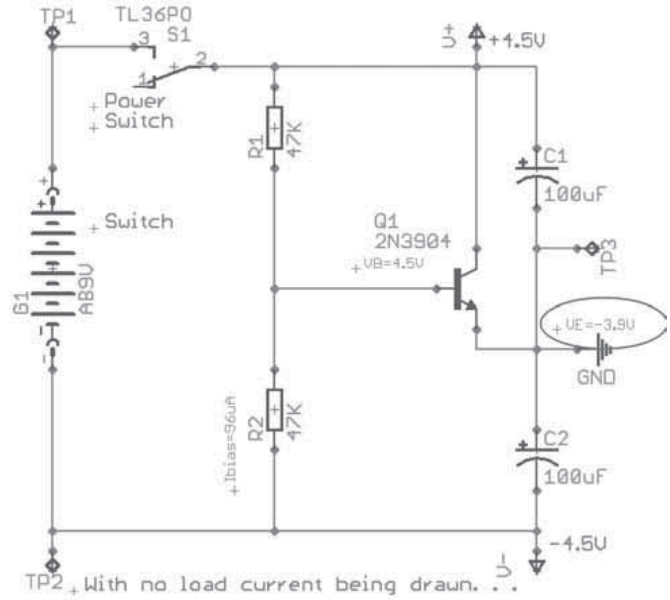


FIGURE 1: Re-drawing of original published schematic.

FIRST

Design concept error. The power supply's artificially centered ground should be capable of both sourcing and sinking current. It doesn't, not really. This design uses an NPN transistor, emitter follower, which can only supply current in one direction. Whether you call it sourcing or sinking depends on whether you think in terms of hole current or electron current.

Either way, this circuit is a one-way street for current and only succeeds due to the large capacitors straddling the output. The lower cap, C2, does CYA (cover your aft-side) duty and effectively provides short-term current for the circuit (load) when Q1 can't provide it (theoretically, when electron current flows from ground to any potential above the voltage of the transistor emitter's artificial ground).

SECOND

This voltage divider bias never centers the output voltage. It is set up for 3.9V output, not 4.5V. For the output voltage (V_E) to be equal to one-half of the 9V provided by the battery, you need to consider the 0.6V drop across the base/emitter junction of the silicon transistor. In order to get 9/2, or 4.5V, at the transistor's emitter, you need to provide 5.1V (1 silicon diode drop (0.6V) higher) at its base. This means, at idle, you will drop 5.1V across R2 and the remainder, 3.9V, across R1. To do this, using values in this range, you should use 39kΩ for R1 and 51kΩ for R2. Now you have 4.5V at the output (V_E) with no load.

THIRD

The regulation of this circuit is not good. With the relatively high values of bias resistors used, and the 2N3904 transistor's beta, the emitter voltage (ground) will theoretically drop from +3.9V to almost zero while supplying a current differential of only 5mA. Not a good thing! A current differential occurs when the current demand between the +V and -V rails is not equal and the difference is what needs to flow through the ground rail.

Fact: The 2N3904 datasheet says at I_C of 10mA, beta $_{(Min)}$ = 100; at I_C of 50mA, beta $_{(Min)}$ = 60. So, at a 10mA current draw, you can only count on a beta of 100 for any given part, unless you test each one and cherry-pick only the good ones.

Fact: The AD712 datasheet states the idling (quiescent) current, $I_{Q\ (max)}$, is 6.8mA. The worst-case scenario is what you must design for, because you can't count on all the parts being average or better. And considering today's availability of counterfeit parts, it's a good chance you may end up with less than prime parts.

Given these facts, let's analyze this circuit with just a small current differential of 5mA.

Using the minimum (worst-case) beta of 100, the transistor would pull 1/100 of 5mA, or 50µA, of additional current through base bias resistor R1. Ohm's law states IR = E; therefore .000050 × 47,000 = 2.35 *additional* volts of drop occur across R1, which will push the base voltage (V_B) down from 4.5V to roughly +2.15V, and the emitter voltage should follow 0.6V lower at +1.55V.

Artificial ground is now at 1.55V above the negative battery rail with the audio signal riding on it piggyback. The so-called "centered" ground is now 2.95V below the intended 4.5V midpoint of the 9V battery.

The three voltage rails (negative, ground, and positive) are now -4.5V, -2.95V, and +4.5V, respectively. If you use the ground as zero reference, then you have a negative rail of -1.55V, a ground rail at 0V, and a positive rail at +7.45V. Not exactly symmetrical, is it? This applies to the sample circuit, as published, while supplying a current differential of only 5mA.

In fact, just 8.3mA of differential current load on this circuit will cause the output to bottom out. That's all it will take to draw another 83µA of current through R1 to cause the voltage at V_B to drop to 0.6V and the output V_E to zero. Ouch!!!

The situation is even worse as the current demand increases. At 50mA of I_C, the 2N3904 beta $_{(Min)}$ drops to 60 or less. You normally won't use this much current, but it reveals a trend that this single transistor and the high-value bias resistors may not be the best design choice for this job.

The circuit author was lucky that he needed to produce only 179mV RMS or so across a 32Ω load to get 1mW of power to the headsets; otherwise, he definitely would have been banging against the negative rail. Obviously, if the bipolar current demand is symmetrically equal, then the sinking and sourcing currents effectively cancel each other and you have no additional current flow through the ground rail because the current differential is zero. However, perfect symmetry is not common unless you are amplifying perfect sine waves or other symmetrical waveforms with no offset voltage and you obtain perfect parts.

A simple, quick-fix for the poor regulation of this circuit: You can stiffen up the bias network by reducing R1 and R2 in value by a factor of ten. This would vastly improve voltage regulation at the slight expense of higher bias current. Make R1 equal to 3900Ω and R2 equal to 5100Ω, and you now have a bias current of 1mA with a perfectly centered output voltage. You can handle 25mA of load current with only a 1V change of V_E. A vast improvement, and the bias current, by itself, would take 500 hours to drain the 9V battery. But, you are still a uni-directional current source.

FOURTH

The amplifier circuit as illustrated on the schematic is a lane changer. It actually switches the left and right audio channels around. Check it out. The tip of the input jack is connected through the op amp to the ring of the output jack and vice versa.

This might be nitpicking, but it could be major for a purist who expects Paul McCartney to always be on his left while listening to a Beatles recording or for any other imaging expectations. I consider this to be a technical faux pas.

In all fairness to the author of the original circuit, I realize that this was a quick, junk box project, and maybe his parts box is even less well stocked than mine. Just the same, a quick disclaimer about the borderline reproducibility of the circuit and that it may need some tweaking would have been appropriate and warned everyone that this was not a highly polished diamond.

THE BETTER APPROACH

If you desire to stay with the one-way current path (a faulty concept) using the output of an NPN transistor, and a CYA capacitor, you can use something like the schematic of **Fig. 2**. Note that this circuit is designed to keep current drain to a minimum while preserving good voltage regulation over a fairly wide load-current delta. The bias current is roughly 1/3 of

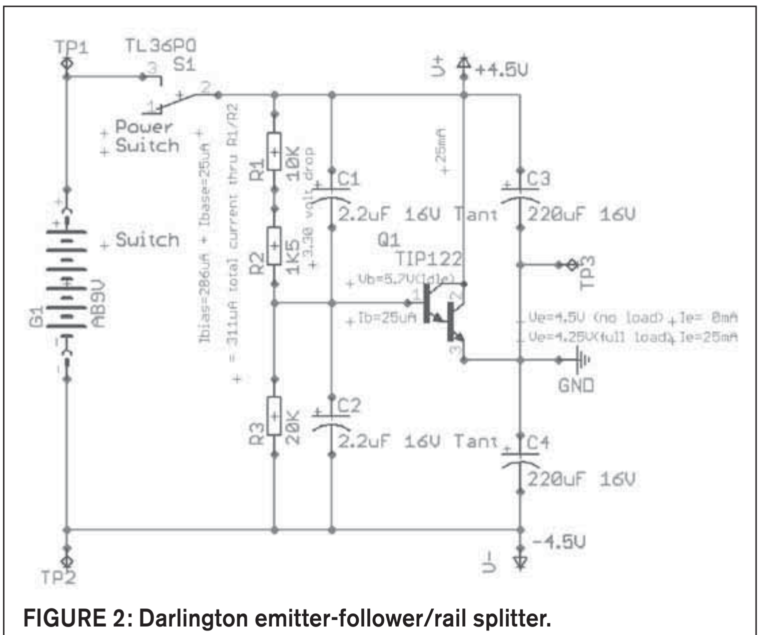


FIGURE 2: Darlington emitter-follower/rail splitter.

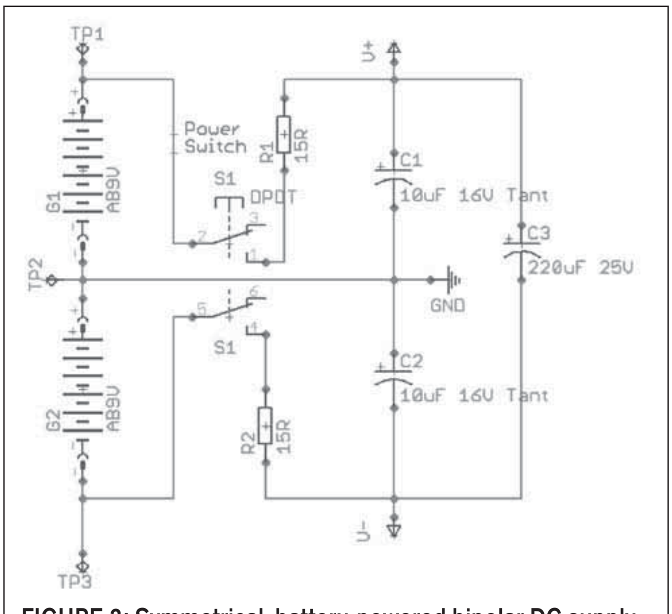


FIGURE 3: Symmetrical, battery-powered bipolar DC supply.

1mA. Remember, again, like the original, this design is not a true, bidirectional current source and is shown simply as a better, voltage-regulator circuit.

Notes: G1 is the 9V battery using S1 as an on/off power switch. R1/R2/R3 forms a precision voltage divider for biasing the base of Darlington transistor Q1 to exactly 5.7V. Capacitors C1/C2 stabilize this base voltage. Q1 has a 1.2V drop between the base and emitter (V_{BE}) because it is a Darlington transistor; therefore, the emitter voltage is 5.7V minus 1.2V, or 4.5V (exact center of 9V).

The TIP122 Darlington transistor has ten times more gain and current than the 2N3904 and has a minimum (worst-case) beta of 1000, so it only draws 25µA of base-current while producing 25mA of emitter-current. So, from no-load to full-load (0-25mA), the small change in base current causes little voltage change in the static 300µA of bias current traveling through R1 bias resistor, thereby maintaining good voltage regulation. Delta V_E = ¼V. If you need better voltage regulation, reduce bias resistor values (R1/R2/R3) by an equal factor. Total bias resistance shouldn't fall below 9kΩ.

Again, even though the typical Beta for a TIP122 is 4,000, you can only use the worst-case specification of 1,000, thus fulfilling your lofty design goal that all parts will work in the circuit.

The circuit's greatest current drain is during turn-on time, when the battery must charge the filter capacitors, which is very brief but intense. You can downsize C3 and C4 to 100µF without any appreciable loss of circuit performance. This also reduces the duration of the turn-on surge. Keep the larger capacitor values if you have a high current demand or must meet large transient demands often.

The best approach is to use the dual battery concept (**Fig. 3**). You can use two 9V transistor batteries for about 20 hours of usable service or two 6V lantern batteries for about 440 hours (18 days) of usable service. Both examples assume a current drain of 25mA and a battery chemistry of alkaline-manganese dioxide. Your mileage may vary depending on the particular circuit that you're powering.

Starting off with ±9V, you will lose some of it in the powered circuit's protection diodes, current limiting resistors, and so on. Additionally, most IC outputs come within only a volt or two of the power rails. Therefore, you are probably dealing with ±6V or so at the load circuit output. If you use a safety margin of 100% headroom, your output will be limited to ±3V peak-to-peak, roughly 1V RMS.

You get a really stiff (low impedance) center point for a ground automatically, as a by-product of the two-battery, series-connection's center tap at TP2, fully capable of current flow in either direction. You can add a pair of current limiting resistors (R1/R2) (2R2 – 15R) to limit the turn-on surge current, charging capacitors (C1/C2/C3), which are themselves mostly gratuitous, but help with the noise and transient current demands.

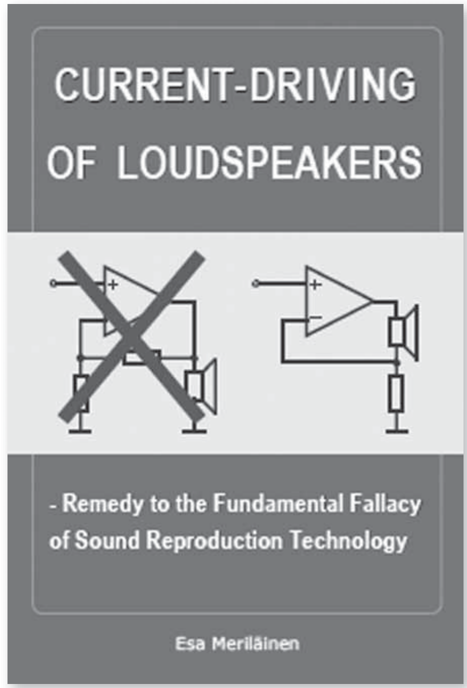
Cheap, clean, and to the point, what more can you ask for? Beauty in simplicity! C1/C2 are tantalum and C3 is aluminum electrolytic. Notice that there is no DC discharge path in this circuit, so there is no bias current overhead when running without a load.

OTHER DESIGN PRACTICES

1. Use worst-case part specifications.
   - Covered in the text.
2. Use commonly available parts (if possible).
   - Don't use rare or specialized parts that may be difficult for others to obtain.
3. Specify where to locate hard-to-find parts.
   - Give the mailing or web addresses for those specialized parts.
4. Avoid using single-sourced parts.
   - Don't use parts that are only available from one manufacturer or distributor unless absolutely necessary, and then think twice about it.
5. Specify necessary parameters when needed.
   - If you need special parameters to be considered, say so. Call out COG, X7R, NPO, Z5U, and other classes of capacitors, if needed; as well as their composition material, such as polystyrene, ceramic, silver-mica, et al.
6. Don't over-specify quality parts if not needed.
   - If you need only 5% or 10% resistors, specify it. If you really need that exotic 0.1% resistor value, then, of course, spell it out.
   - Keep the parts list friendly. Others may not stock 10,000 1% resistors.
   - If you must use mumbo-jumbo, essence-of-the-ether capacitors, say why, or give some alternatives. Most of us can't hear the difference anyway.
7. Be mindful of electrolytic capacitor voltage ratings.
   - Keep them high enough to be safely above any voltage that they would be exposed to, but not much more than 50% above that highest voltage.
8. Be extra careful when designing circuits that involve working directly with the power mains.
   - Spell out the dangers when working with lethal voltages or currents.
9. Try to be clairvoyant about the worst-case conditions that your circuit will be exposed to. Sooner or later, it will!
   - High line voltages, low line voltages
   - Shorted outputs and over-driven inputs
   - Line voltage spikes and surges
   - RFI sources
   - Stray magnetic sources
   - Temperature and humidity extremes
   - Children, pets, and fools *aX*

Current-Driving of Loudspeakers

Reviewed By Joseph DeMarinis



In his book *Current-Driving of Loudspeakers* (available for $34.95 from Old Colony Sound Lab, www.audioxpress.com), subtitled "Eliminating Major Distortion and Interference Effects by the Physically Correct Operation Method," author Esa Meriläinen challenges the conventional wisdom of driving loudspeakers from amplifiers that are voltage sources. He is clearly a very skilled and knowledgeable (Master's degree) electrical engineer, resides in Finland, and has written this book as the result of a long-term hobby interest in the subject. Little else is known about him. His 342-page treatise is very thorough and touches on just about every aspect of sound reproduction. It is an excellent tutorial, leaving no stone unturned.

Because it is well known that the *current* passing through a loudspeaker voice coil is what causes it to move, the idea of tightly controlling that current (rather than the voltage across the coil) is a very seductive topic. Current drive amplifiers have been possible since the invention of negative feedback in 1927. I am sure that over the intervening 83 years many audio equipment designers, in quest of a competitive advantage, have experimented with the subject. Yet, the practice has never been adopted by the industry. The author is well aware of that and has poured his heart and soul and his engineering skill into trying to convince us all that we have gone down the wrong path.

This is not light reading! The book is very heavy with math and is an excellent refresher course in the mathematics of circuit and electromechanical system behavior. Whether you agree or disagree with his conclusions, the book is so detailed and thorough that it deserves serious consideration.

However, the bottom line is whether current drive truly results in improvements, and I was compelled to find out. I modified one channel of an old stereo receiver to be a current source and ran a bunch of lab measurements and listening tests on a few available speakers. It was a much more complex effort than I'd anticipated. More about that later.

An ideal voltage source delivers a voltage that is held constant regardless of the load. In an ideal case, its source impedance is zero (looks like a short circuit). An ideal current source delivers a current that is held constant regardless of the load. In an ideal case, its source impedance is infinite (looks like an open circuit). Bear in mind, however, that real voltage and current sources don't achieve those ideals.

According to conventional wisdom, there are two very good reasons why voltage drive is used universally: loudspeaker resonance and damping.

Both the efficiency and impedance of a loudspeaker increase greatly at frequencies near its resonance. When driven by a constant voltage, the increased impedance causes a reduction of the current in the voice coil, so the system is approximately self-compensating. However, if the current is kept constant, there would be an enormously loud audio response peak at and near system resonance, because of the dual effect of higher efficiency and the increased power a current-drive system will deliver into the higher impedance near resonance.

Because a loudspeaker system is basically a mass and a spring, once set in motion it will continue to oscillate on its own, unless there is some damping mechanism to absorb that kinetic energy. Some of that damping is done by the air pressure load, design of the cone suspension, and the stuffing inside the enclosure. But much of the damping is electrical. When the coil moves, it acts as a generator. Because a modern voltage-drive amplifier has a very low source impedance, it acts as a short circuit load on the voice coil and quickly suppresses the unwanted motion.

This describes the present conventional wisdom. The author of this book does address those issues. He proposes greatly reducing the Q of the mechanical resonance, plus electronic equalization to compensate for any remaining resonance peak. He claims that electrical damping by a voltage-drive amplifier is ineffective at frequencies other than resonance and that reduction of the mechanical Q provides sufficient damping.

This implies that loudspeakers designed for voltage drive won't necessarily work with current drive and that the required electrical equalization must be carefully matched to the speaker(s). That can become quite complicated with multi-driver speaker systems and is a strong incentive to include equalization in the built-in power amplifier or crossover network. In other words, implementation of these ideas requires a system approach, as can be done effectively with *powered loudspeaker products.*

The subjects mentioned here are those that immediately hit an informed reader "between the eyes." This 342-page book contains a lot more!

THE BOOK ITSELF

In the preface the author makes claims

that remind me of typical Audio advertising fluff. In chapter one, he compares great electrical battles, such as between Edison and Tesla, AM versus FM radio, and so on. After reading that, I was anxious to get into the meat of his case. But not so fast. . . Chapter two is an extensive math lesson dealing with *Second Order Systems*. In the back of the book, appendices A through E are tutorials dealing with *Complex Numbers*, *Linear Systems*, *Frequency Content of Signals*, and *Differential Equations*. Appendices F through H are patterns for PCB copper. In chapters 10, 12, and 13 he presents designs for an amplifier and special measurement devices. Chapter 11 discusses *Loudspeaker Implementations*. So, buried amid all the math are some practical "how to" instructions.

But the chapters that fascinated me the most are #3, *Operation of the Electromagnetic Transducer*, and #7, *Modeling & Simulation* (of loudspeakers). Those are great analyses of the deep-down fundamentals of loudspeaker operation in which he reveals all manner of problems and reasons loudspeakers don't work very well. Some relate to drive considerations, some are more general in nature. Those chapters enticed me into running experiments to investigate certain phenomena.

MAJOR ISSUES

Most of his objections to voltage drive stem from the back-EMF *(Electro-Motive Force is another way of saying voltage)* that is the result of the motion of the voice coil in its magnetic field. (Remember when you studied electric motors?) With a low impedance (voltage) source, that EMF causes reverse currents to flow in the voice coil. In an ideal case, those currents would be exactly out of phase with the driving current, would be part of the resistive component of the speaker's motional impedance, and would do no harm. However, the motion of the cone and therefore the back-EMF is usually phase shifted from the drive, may be nonlinear, and may also be due to extraneous forces on the cone, such as pressure waves reflected from the rear enclosure. So there is potential for distortion. With a current-drive system, that back-EMF sees an open circuit and no current will flow.

However, you will see that those "unwanted" currents can also play an important role in suppressing spurious cone movement. Are they good or bad? Perhaps the answer to both questions is "yes."

THE AMPLIFIER

To get started I modified the left channel of an old Onkyo 40W stereo receiver to achieve current drive. I sampled the output current through a 0.56Ω resistor in series with the speaker load and used that as the source of negative feedback, instead of the output voltage. I adjusted the amount of current feedback so that both channels delivered the same power into a 10Ω load, from the same input signal. That enabled me to do monaural A-B comparisons with a DPDT switch.

Initially the amp took off at about 1.5MHz and would ring with an inductive load, when hit with a fast pulse. Fixing that required routine engineering, and, when done, I verified that both amplifiers had the same clean transients and very flat frequency response over the audio spectrum. The source impedance of the voltage amplifier is about 0.33Ω. The current source impedance is about 1150Ω.

There are a few effects you notice right away when using a current amplifier. One is that the output voltage skyrockets when the load impedance increases (as with speaker resonance) and it is very easy to run it into clipping. When the output voltage is clipping, there is no control of the current. So you must continuously monitor voltage clipping.

With a partially inductive load, the rise and fall time of current pulses and the subsequent audio response are much steeper with current drive. That steeper rise is driven by voltage spikes. So, once again, considerable headroom is needed with regard to the voltage capability of the amplifier. Understand that the current in the load always comes about because the amplifier delivers a voltage. What we call current drive is, in reality, a voltage source in which the voltage is automatically and instantaneously adjusted up or down, to maintain the desired current.

You should not operate a current amp with the load disconnected. The output voltage rises with an incredible gain and hits the ± rails very hard. That can damage the output transistors. Similarly, you

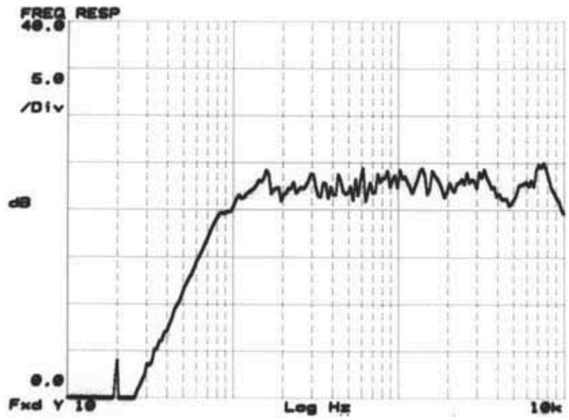


FIGURE 1: Aura, voltage drive.

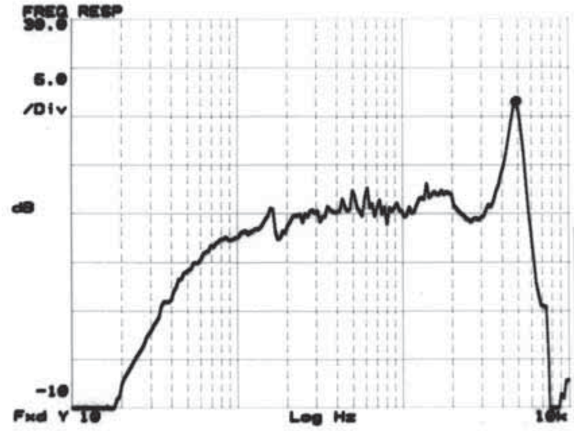


FIGURE 3: SEAS, voltage drive.

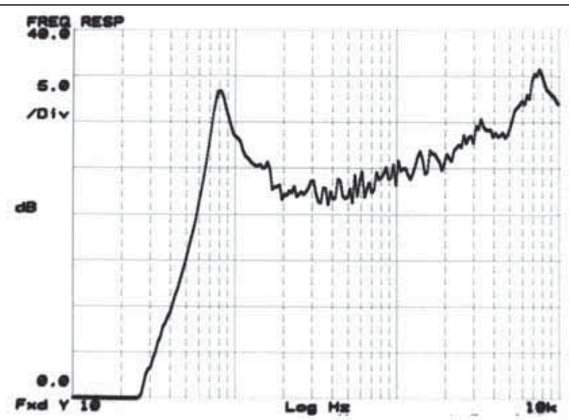


FIGURE 2: Aura, current drive.

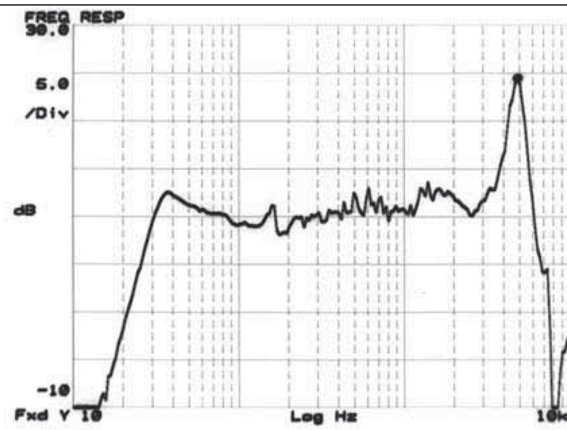


FIGURE 4: SEAS, current drive.

can damage a voltage amp by running it into a short circuit.

Also realize that while a voltage amplifier will deliver higher power into a low impedance load, a current amplifier delivers higher power into a high impedance load. With current drive, a 4Ω speaker receives less power than an 8Ω speaker. That may seem counterintuitive because you are so accustomed to dealing with voltage sources, be they amplifiers or household electricity. Remember that power = I^2*R.

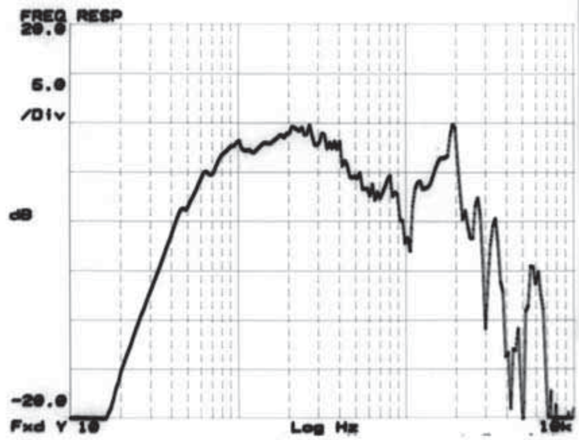


FIGURE 5: KLH woofer, voltage drive.

TESTS AND EXPERIMENTS

Using the modified amplifier, I ran several tests on three different speakers to see how they reacted to voltage or current drive stimulus. The speakers were an old KLH (Model 17) bookshelf system, a SEAS 5.5″ woofer (W15CY-001-E0015) in a sealed enclosure, and an Aura 3″ (NS3-193-8A) wide range transducer also in a sealed box. In the case of the KLH, the mike was in front of the woofer, so ignore the high end of those curves. I made no attempt to modify or equalize the speakers.

The first test compared frequency response with voltage and current drive. As expected, current drive made the deep bass louder and the treble more crisp. The increased bass was generally pleasant (people do love boosted bass!). The crisper highs were a mixed bag, depending on the source material. A complete design would include damping and equalization

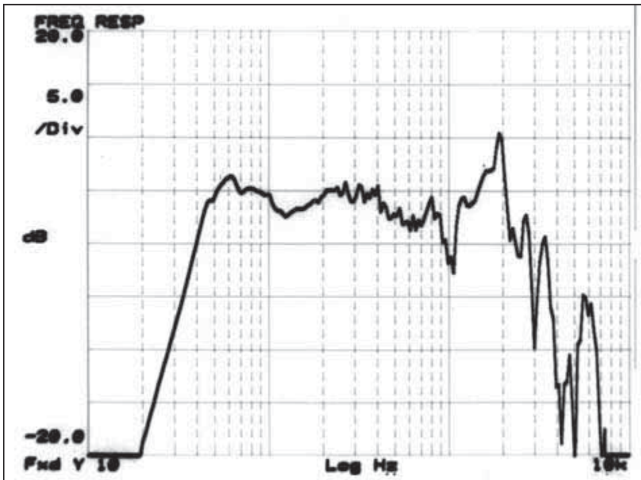


FIGURE 6: KLH woofer, current drive.

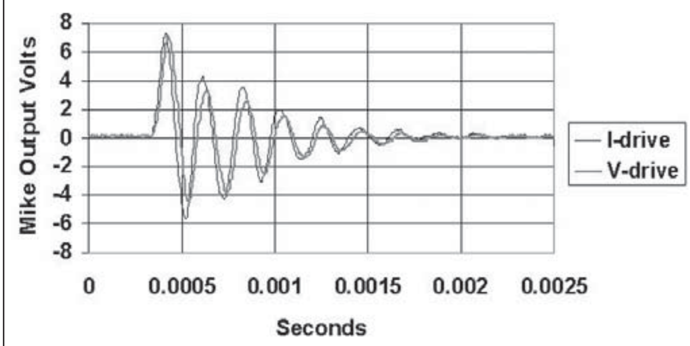


FIGURE 8: SEAS pulse responses.

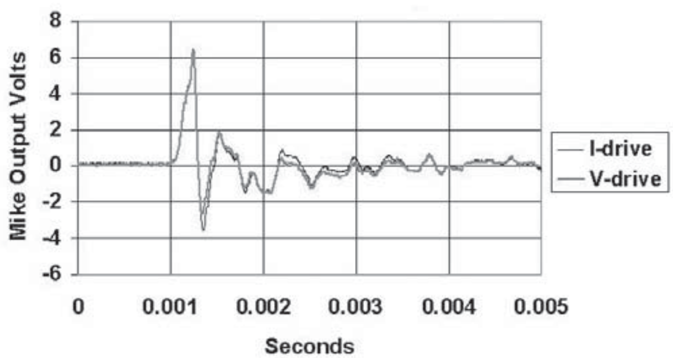


FIGURE 7: KLH pulse.

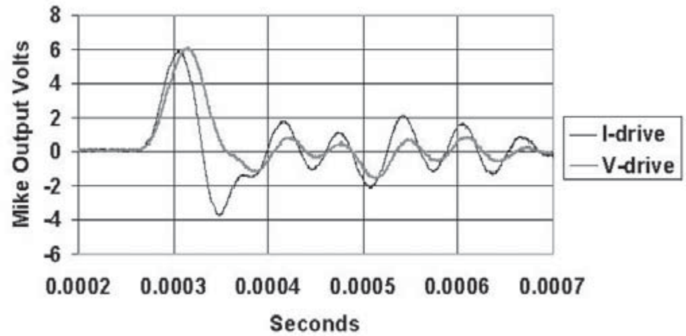


FIGURE 9: Aura pulse.

to render both ends of the spectrum to be flat. The six swept-sine response measurements are shown in **Figs. 1-6**.

As you can see, the different speakers responded to the change in different ways. In two of the cases there was a significant benefit at the low end. But the Aura is clearly designed to behave well with voltage drive and doesn't like current drive at all.

The next test was to hit those speakers with a 600ms pulse and look at how their acoustic responses are damped. That pulse is long enough to get the entire cone structure moving, but not long enough to interfere with the subsequent response (**Figs. 7-9**).

Bear in mind that when hit with a transient, the cone will continue to vibrate (as does a drum). That, of course, is a distortion that can cause a "muddy" sound. So the quicker that movement is damped, the better.

Here you see that with the KLH and SEAS drivers there isn't much difference between voltage and current drive damping. For the Aura, however, the voltage-drive amplifier damps it significantly quicker and very effectively. The

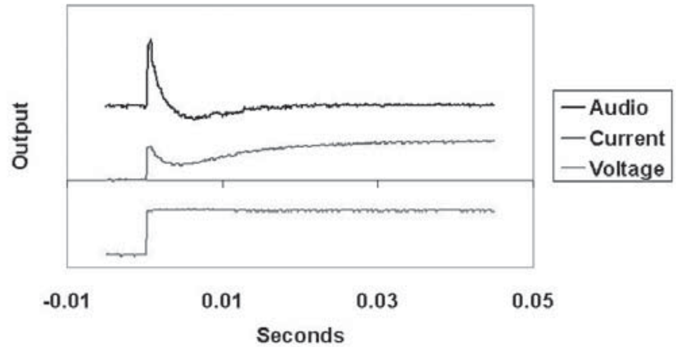


FIGURE 10: SEAS DC power on.

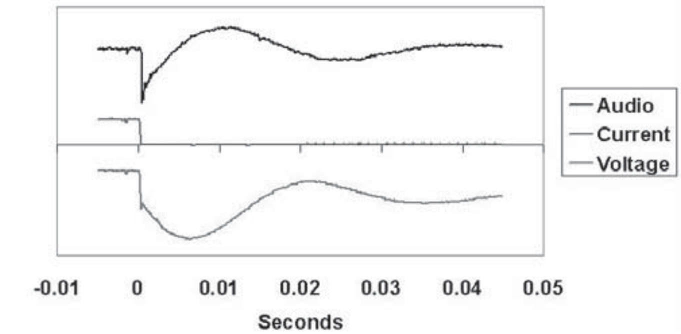


FIGURE 11: SEAS power off.

frequency response data as well as impedance measurements (not shown here) indicate that the Aura has a much higher Q than the other two speakers. This tends to support the author's claim that controlling the mechanical damping factor is effective, and, if done properly, electrical damping may not be necessary. For me, this was a very revealing exercise, but just one aspect of overall performance.

To illustrate the existence of the back-EMF and resulting currents, I connected the terminals of the SEAS woofer to a well-regulated DC supply and then broke the connection, while my oscilloscope and B&K measurement mike "watched" what happened during the make and break. Each of the charts in **Figs. 10** and **11**, reading from the top down, shows the audio response, the current, and the voltage.

Notice that when the very low-impedance power source is applied, there is an immediate audio response, but subsequent cone movement is critically damped. While that is happening, the cone motion produces a back-EMF and the "reverse-current," which reduces the net input current (just as in an electric motor). The "reverse current" does the damping. The externally applied and well-regulated terminal voltage is immovable. This illustrates a voltage-drive situation.

When the power source is disconnected, the voice coil is now looking into an open circuit. There is an immediate audio response as the cone returns to center, but then it continues to move as indicated by the audio ringing that follows. Note that the EMF caused by the cone motion appears at the speaker terminals. Because that is now an open circuit, there is no current and no power transfer; therefore, no electrical damping. Note also that the acoustic output and back-EMF are not exactly out-of-phase. That is in part because the back-EMF represents the velocity of the voice coil and not its position. Also, the interaction between the cone and air is not a simple matter, as the author attempts to explain in chapter 3, pages 41 and 42. Air, too, has mass and elasticity.

The "Power ON" response is similar to the behavior of the earlier pulse experiment. Even with current drive, you don't see the large "Power OFF" oscillation when the pulse is over, in part because the amplifier is still connected. Even the current-drive amp has a finite source impedance and does some damping.

I must emphasize that this DC experiment is a "science demonstration" and will not occur in normal use.

Now, if you think that current drive (or voltage drive) will provide an iron grip on the cone movement, think again. The current provides the force. But the cone reacts to that force with a mind of its own. The author takes you through models of that behavior in great detail, involving electrical resistance and inductance of the voice coil, the motional impedance of mass, elasticity, and damping, and the interaction with air. The effect of all that is revealed when you look at the phase relationship of acoustic output versus input voltage or current, over the frequency range of a speaker (**Fig. 12**).

These are measurements made on the 5.5″ SEAS woofer. The thick lines represent the acoustic output and the thin one is the electrical phase of input current. In both cases, the phase reference is the voltage. At resonance (about 36Hz) the voltage and current are in-phase, as would be expected with a parallel tuned circuit. But the acoustic output in that region is nearly 180° out-of-phase, becomes exactly out-of-phase below reso-

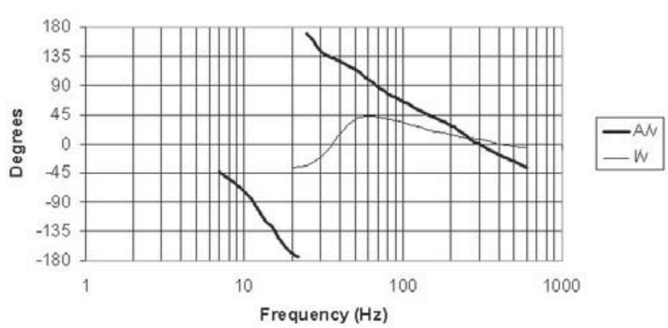


FIGURE 12: SEAS phase data.

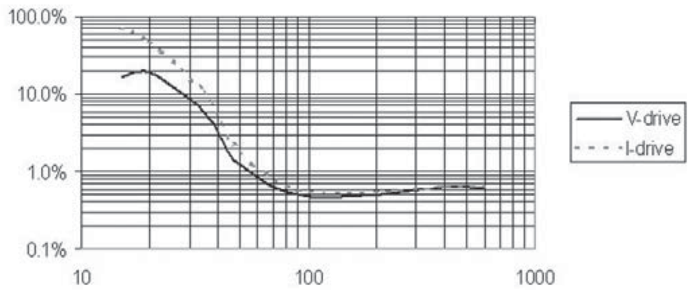


FIGURE 13: SEAS distortion.

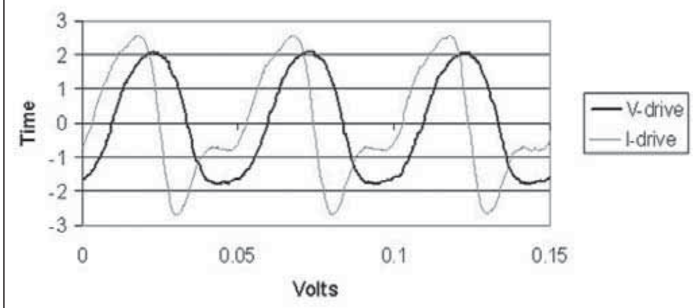


FIGURE 14: SEAS 20Hz.

nance, and drifts back in-phase as the frequency decreases or increases. This illustrates the point that neither the voltage nor current is directly controlling diaphragm motion and air handling.

DISTORTION

All of this may or may not be interesting, but the reason for this journey is to attempt to reduce distortion. So, using low power (less than 0.2W) that stressed neither the amplifier nor speaker, I measured acoustic distortion over the useful range of the SEAS woofer. The result is shown in **Fig. 13**.

The heavy line is with voltage drive and the dotted line is with current drive. You can see that distortion is quite low at the mid-range, with no significant difference between voltage and current drive. But it increases rapidly as frequency drops and cone excursion increases. Bear in mind that below resonance it is virtually impossible to get a really clean acoustic sine wave. Nonetheless, in that region, voltage drive seems to exert better control of the cone. The plot in **Fig. 14** shows the SEAS acoustic waveforms at 20Hz.

It is interesting that when I took this data, the distortion of the input current was 0.5% with current drive and about 9.6% with voltage drive. It seems that the reverse current generated by spurious cone movement "distorted" the voltage drive net current in such a way as to correct some of the distortion.

I performed a similar acoustic distortion measurement of the KLH system at 18Hz, at the same power level. The distortion was 12% with voltage drive and 19% with current drive.

I have been in e-mail contact with the author and shared this finding for his comment. He was not surprised and pointed out that the harmonics of very low frequency signals can fall in the range near mechanical resonance and would naturally be exaggerated. He believes that reduction of the Q would help and also made the point that this may be a moot issue because there is very little sound content to be found at those low frequencies.

In these tests, the KLH, which has the lower Q, had less distortion than the SEAS. I've also watched the real-time spectrum of music on the FM band and have not seen anything below about 30Hz.

CONCLUSION

This investigation has been limited in scope because I could not modify the loudspeakers. I compare this to a weight-loss plan. To achieve the results, you must do the whole program. Therefore, I am going to resist the temptation to draw any summary conclusions. Let the reader decide.

The book's author points out many problems associated with voltage drive, and my data has pointed to some problems with current drive. But today's powerful computer modeling and systems approaches, as well as intense competitive pressure for product differentiation, could yield some high-end systems based on current drive. I would like to see that happen! *aX*

Emotiva XPA-2 Power Amplifier

By Tom Lyle

	Emotiva Audio Corporation
	131 Southeast Parkway Court
	Franklin, TN 37064
	(877) 366-8324
	www.emotiva.com
	$799

I've been involved with audio for quite some time, and I'm having quite a bit of trouble recalling a power amplifier that has this much power. The Emotiva XPA-2 two-channel power amplifier's specified output is 250W per channel into 4Ω, 500W per channel into 8, and its power *doubles* when used in bridged mode. It costs $799.

But you must be wondering the same thing I was when I first heard about the XPA-2: What kind of amp can this relatively small amount of money buy? Actually, it is an *infinitesimally* small amount of money when you consider what other companies ask for a 250W per channel amplifier. And probably an even more important question is: How will it stand up to other power amplifiers with this much power?

FEATURES

After nearly straining my back and legs lifting the 75 lb 17" × 7.75" × 19" amplifier out of its box, then thumbing through the XPA-2's manual, I can't see how they managed to not only stuff all this technology into this massive cabinet, but also sell it for a mere $799. There are tons of new and some not-so-new companies producing equipment in China and the Far East that might cause you to pause and think about the factory workers who might be earning only pennies an hour to assemble these components for us spoiled audiophiles. But it is not my job to ponder such matters in this forum; I just listen to the finished products.

After listening to the Emotiva XPA-2 power amp, I can tell you that this spoiled American audiophile was mighty impressed with the results. I wouldn't recommend replacing your multi-thousand dollar American-made amplifier with this unit, but for those who need plenty of *power* but might not have the scratch to make that kind of investment, this amp will do just fine.

If you shop around, you could easily find an amp with this much power for this much money, but you'd likely end up with an amp with either a Class D or H switching power supply, both of which are very common for pro audio applications, such as public address and disc jockey systems. But Emotiva's beast is a fully discrete Class A/B amp that uses 24 high-current output devices. Emotiva claims that their design achieves the "utmost in neutrality" with "detailed highs and a powerful, tight, extended low-end" with an "open soundstage" with "great resolution, detail, imaging, and separation."

Of course, you've all heard these types of claims from audio companies about their beloved products, but Emotiva has certainly packed the XPA-2 with impressive internal components such as a huge toroidal transformer, glass epoxy printed circuit boards, 1% tolerance resistors, and metal film capacitors. The amp has soft-start circuitry to prevent spikes on startup, and a modular channel design that, according to the company, gives rise to greater channel separation. This nice-looking unit has all the functionality that you would expect from an amplifier manufactured for use by audiophiles.

The ¼"-thick steel black front panel has silver accents on its edges and a large cylindrical silver-colored power switch (which changes the amp from standby) and sports nifty switchable blue LED power meters that illuminate in a horizontal row across the top of the front panel (check out the Emotiva website to see the meters in action). The back panel is loaded with fairly common features for a large two-channel amp: In the upper left-hand corner a 9 to 12V trigger input, and switches for both the power and power meter LEDs; and to the right of those and occupying the center of the upper portion of the panel are the inputs—you can select either balanced XLR or unbalanced RCAs with a toggle switch. On the lower portion of the back panel are the hefty speaker binding posts, an IEC input for its detachable power cord, the main power switch, and a locked 115/230V AC selector.

LISTENING

Despite its size and weight, I maneuvered the Emotiva XPA-2 onto the second shelf of an Arcici Suspense equipment shelf without too much trouble. The preamp used for most of the review was the tubed Balanced Audio Technology VK-3iX, but often I used either the tubed Acousticbuoy Scorpio or the solid-state Burson Audio PRE-160 preamp. The digital front end consisted of either an Arcam DiVA CD192 CD player or an Oppo DV-980H universal player as a transport feeding the Acousticbuoy DAC2488 V3 digital-to-analog converter, and I also used the Oppo's analog outputs when occasionally spinning SACDs. The analog front end consisted of a Basis Debut turntable with a Lyra Helikon cartridge mounted on a Tri-Planar VII tonearm, with either a Pass Labs XP-15, conrad-johnson TEA2, or Rossner & Sohn Canofer phono preamp. The speakers were the Sound Lab DynaStat electrostatic/dynamic hybrids assisted by a Velodyne HGS-15b subwoofer.

As expected, the 250W per channel that the Emotiva XPA-2 power amplifier provided was plenty of power to drive the Sound Lab speakers. I've used very powerful amps with these speakers before, and I've found that although they are acceptable, anything less than 200W per channel is not enough to bring out the very best that these speakers offer. But the XPA-2 was more powerful sounding than I expected, and was subjectively louder than my 250W per channel Krell that I usually use in this system. But it was more than just loud; you could call it muscular, brutish, brawny, and so on.

> "The Emotiva XPA-2 is one of the greatest bargains in audiophile power amplification today."

Even more important, this amp had enough macrodynamic slam to stand with any amp at any price when judged on this trait alone. I might be getting ahead of myself a bit here, but, as with lesser priced amps, it didn't have the kind of detailed layering of sounds and other nuances, including a totally grainless treble. These qualities set the higher price units apart from the lesser contenders. Don't get me wrong; I think the XPA-2 is a magnificent bargain. Just don't directly compare it to an amp costing four times as much.

SOMETHING

When I hit play on The Beatles' remastered version of *Abbey Road*, the recording was laid bare before me, and I was reminded that I'm still not used to it sounding this good on CD. The bass frequencies on the CD reached subterranean levels and were able to shake my bones, so when I cranked up the volume during Ringo's drum solo on "The End" it sounded as though I was hearing it for the first time. Even though this album is undoubtedly a treat for all who have suffered through 25 years of digital neglect, whether this album is eligible for inclusion into any audiophile categories is debatable.

Nevertheless, the XPA-2 did a great job of unwrapping all the details that were hidden (or just sounded bad) on the original digital version. And even though it doesn't take an extravagant system to hear the differences (the improvements over the original CD issue in 1987 are so profound that even a system recommended by *Consumer Reports* will reveal them) the XPA-2 made this great album fun to listen to. Beatles fans could easily use this amp as a tool for meticulous sonic analysis, and at the same time risk breaking their apartment leases.

Although the term "soundstage" should be reserved for real instruments recorded in a real space, there is a certain amount of front-to-back layering that I've come to expect from the best recordings played on decent equipment, regardless of their source or genre. The eight-track origins of this recording—when played back even on the best equipment—render it pretty much a two-dimensional affair, and through the XPA-2 this album suffers even more. The XPA-2 could pull off the illusion of imaging just fine, so even if its portrayal of the lateral distance between vocals and instruments is impressive, the depth component of these images was somewhat attenuated.

This led to a lack of air around the instruments that some other, more refined amps seem to be able to draw out of the recording. Instruments on the recording had a lifelike persona, such as George's acoustic on "Here Comes The Sun," but for the most part I was left with enjoying this album for what it is—a great band recording great songs. And the XPA-2 transferred this fact to the speakers in a somewhat honest way; there was no coloring of the sound, and for that this amp scores very highly. So I'm probably making more of this than I should, because otherwise I was still mightily impressed, and not just because of the XPA-2's brawn.

ORCHESTRA SOUNDS

Turning to Ravel's *Daphne et Chloe* with Pierre Boulez conducting the Berlin Philharmonic on DG, the XPA-2 showed that it could easily deal with the shifting orchestral swells and ebbs of the lush scoring. This is a recording of a huge orchestra plus a chorus, and the XPA-2 effortlessly took to the climaxes of the piece and pumped out the watts to the speakers. The transient response of this amp is also very impressive. There are occasions where the percussion is front and center, and any instrument in the battery of percussion—which includes timpani, tam-tam, triangle, snare or bass drum, castanets, tambourine, and cymbals—had a natural transient response that was balanced and unaffected, and was neither etched nor slow to react. It also displayed its explosive macrodynamic response on this piece; the right side of the orchestra sounded as though it was going to burst into the room at any second.

When I turned to a favorite vinyl edition of the same piece, the Living Stereo version of Charles Munch conducting the BSO on RCA/Classic Records, I had the same positive experiences as when playing the CD. While it was a completely different interpretation of this piece—the Boulez having more of an international flavor compared to the "French-ness" that Munch injects with his forces—there was still the volatile mix of quiet impressionism with explosive outbursts of the wordless chorus and driving orchestral crescendos that were easily handled by the Emotiva. But again, the lack of vivacity that comes from the inner detailing of the orchestra was there, along with the subjective sense that the overall sound is not as refined as with a more expensive amp.

Even if my complaints regarding the sound of the XPA-2 are in reference to its subtractive faults, this doesn't apply to its extension of the frequency extremes, nor did it have a recessed midrange. But there was a lack of lifelike sparkle to its sound, as though there was a decrease in its dynamic distance, that is, the difference in dynamic range between the loudest and softest sounds that are playing simultaneously. And in combination with its lack in the complexity of the layering of the sounds within its soundstage, this ended up being a mixed bag—plenty of power, transparency, and extension, but lacking in sonic refinement.

CONCLUSION

The Emotiva XPA-2 is one of the greatest bargains in audiophile power amplification today. If you were to look for an amp with this amount of power with the internal features of this Class A/B amp you would be hard pressed to find anything *anywhere* near its price that would come close to its level of performance. While it can't compete with amplifiers five times its price, that is because that's not a fair comparison. However, if you want an amp with a *huge* amount of power and an uncolored sound, and can't spare the money for an American-made muscle amp, the XPA-2 is definitely worth considering. I can just imagine a stack of five of these babies bridged to mono in a family's home-theater equipment closet, and only having to fork over the equivalent of one "average priced" boutique amplifier. Batten down the hatches! ***rr***

PDRS

Thank you for the excellent review by Keith Levkoff about the Personal Digital Recorders (March and April 2010). This two-part review is very accurate and revealed a completely new world (at least for me). I didn't even know that such devices existed and, according to the review and to several comments on the web, PDRs are capable of very high level performance. Missing from the review, however, were the distortion and noise figures, which I consider paramount for this quality gear. By searching H2 ZOOM on the web, I found confirmation of all the technical data mentioned in the review, but no distortion or noise figures.

I also wrote to their info link (info@zoom.co.jp) and was partially successful. In one day they answered with the following "essential" message: "Regarding the THD of the H2, Internal microphone: 0.02%, External microphone input: 0.01%, Line input: 0.05%." It looks as though THD is lower, but I am not sure. Probably the figures are typical values or, as they pointed out, the figures include the noise (a frequent practice). I just wanted to share this with the other readers—not a great deal of information, but better than nothing.

Finally, I think in a short time I'll be the owner of a H2 Zoom anyhow!

Enrico Figurelli
efig.1@alice.it

Keith Levkoff responds:

I, too, was surprised at how far these devices have come, and how useful the current ones can be. That said, I'm very dubious of the numbers they quoted you.

The trend I've noticed among both manufacturers' claims and reviews of PDRs is to avoid "hard numbers." Most reviews treat them as a musician would look at a low-cost microphone: they might include sound samples but they won't bother to take measurements. They don't expect measured accuracy, but instead look for "good sound at the price." (The Zoom H2 includes four microphones in a device that retails for as little as $149; the microphones themselves cannot cost more than a very few dollars each wholesale. I simply cannot imagine that they are especially flat, quiet, or low in distortion. If Zoom had used fancy DSP equalization techniques, their marketing department would be talking about it.)

Another tendency of manufacturers of non-audiophile digital electronics is to quote theoretical numbers. Computer sound cards often tout 96/24 capabilities, followed by THD and noise figures that are obviously the theoretical limits of the sample rate and bit depth. Other times they will quote numbers that are clearly taken from the manufacturers' spec sheets for the DACs or op amps they use. I doubt that they come anywhere close to that performance running inside of a computer, sharing power and ground with the rest of the computer, and with no shielding whatsoever. (A good op amp today can achieve ridiculously low levels of distortion, but even audiophile preamps with fancy power supplies never get anywhere close to them in the real world.)

In the case of the Zoom, I really doubt that those numbers they provided you are anything more than the numbers from the components they used (I *could* be wrong)—otherwise, they would at least be mentioned in the literature. I'm not equipped to take measurements, and none of the reviews I could find did either. So, we're sort of stuck with deductions (until someone provides measurements).

The internal microphones on the Zoom are tiny low-cost condenser elements. Typically, such units are more or less flat to 13kHz or possibly 15kHz (most of the ones I've seen specified seem to have a hump near the top of that range followed by a sharp drop-off). I seemed to notice a slight loss of very low bass on organ music, but it was easily fixed by a little bit of EQ. Typically, in this type of device, the line inputs go almost straight to the DAC chip, after an attenuator, and so are probably quite flat and quiet. Zoom doesn't talk about circuit details at all (and, from pictures I've seen, the board is very complicated to try and trace), but some digital attenuators are intrusive.

When using line-level inputs it would probably be best to set the attenuator in the Zoom all the way up and use the output level control of your source (if it has one) to match levels. From my experience, and a few comments I've read here and there, the microphone preamp in the Zoom isn't especially quiet, so that strategy wouldn't be a good idea there. I've only ever used mine with the internal microphones and an external Edirol stereo unit—and the noise floor was acceptable (but noticeable) in a very quiet recording. You definitely want to get those pads set right before you use the level control, and try to record at a reasonable level.

Of course, in a live recording, crowd noises are always much louder than the noise floor of the preamps anyway; and in a studio environment you would be able to set the levels more carefully to avoid recording at -24dB to begin with.

In the context of LPs, and at the risk of offending a few folks, absolute flatness and ridiculously low THD and noise aren't really factors there either. I'm pretty sure that the Zoom's line inputs are flatter, quieter, and have lower distortion than any combination of record lathe, vinyl, turntable, and preamp out there. If the circuitry is relatively straightforward, there is no reason frequency response should be anything other than ruler flat with a downturn at each end.

Digital distortion (or noise) is definitely more annoying than analog hiss, and so we must ensure that it never becomes audible. As long as the digital noise floor remains at least 10 or 20dB below the analog noise floor, however, the analog hiss serves as a dithering signal and totally prevents the digital noise from being audible (this is not just a simple masking effect—dithering actually prevents the digital artifacts from occurring). In the real world, no analog source is likely to approach the signal-to-noise performance of even a "modest" digital device such as the Zoom as long as the levels are properly matched.

For those who are interested, my Edirol R-09HR claims a frequency response of "20Hz to 40kHz ±2dB"—presumably at the 96kHz rate. They also specify input and output impedances. They neglect, however, to specify S/N or THD. (The Edirol costs about $299—twice as much as the Zoom H2. Zoom also has a new larger model with more features—the Zoom H4n, which is worth checking out. There are lots of other alternatives as well.)

AMP RESPONSE

Kent Smith's interesting article (*audioXpress* May 2010, p. 20) comparing a solid-state amplifier against a tube amp by comparing waveforms raises a question or two. Briefly, his tube amp (Fig. 1, on the left) shows an ideal time adjustment between input (CH4, green) and output (CH3, red) at high frequencies, as are represented by that brief impulse at 0.8ms.

But later in those waveforms, at, say, 5ms, where they are much smoother and where the low frequencies dominate, the output slope crosses zero 0.2ms sooner than the input, as if low-frequency phase lead might be present. In his solid-state waveform, the high-frequency time adjustment again is good, and now the output is more closely aligned with the input even at 5ms.

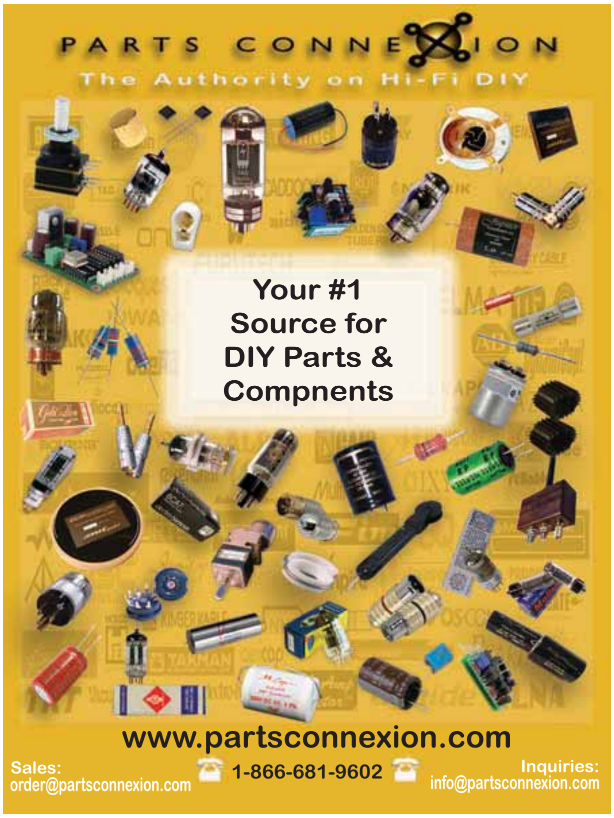
Has Kent measured the low-frequency amplitude and phase responses of his two amps? And could that explain the waveform discrepancy?

Mike Lampton
Berkeley, Calif.

Kent Smith responds:

I did measure the low-frequency response of both amplifiers, because I was a little afraid the response of the chip amp may not be where I thought it was. However, the chip amp was flat down to below 20Hz. The tube amp was flat to near 30Hz, the OPT rating. I have not tried phase response, but I'm sure the transformer in the tube amp would look worse. Fortunately, there just doesn't seem to be much music material down in that region, and the drum in the plot is near 120Hz, well away from the rolloff region. aX

Audio Marketplace



CONTRIBUTORS

Dick Crawford ("The BTO," p. 5) has been building hi–fi amplifiers since the Williamson, which was a Class A design of the 50s. In the interim, he has earned a BSEE from Cal Poly (SLO), and an MSEE from Stanford. He works as an analog design engineer in the San Francisco area.

Kenneth L. Bird ("The Fire Stick Speakers," p. 10) has been building speakers since 1954, and was installing 12″ speakers in the rear decks of 1958 Chevys and Cadillacs long before the word "autosound" was coined. Armed with a BSEE and much later with an MBA, he has spent most of his career in audio related pursuits with such firms as GTE, Rockwell International, and Shure Brothers. Currently, he is a Product Manager with MAXTEC International, makers of the B+K line of test equipment.

Bard Kallestad's ("Designing for Everyone," p. 14) 40-year electronics association includes work in commercial radio broadcast engineering, electronic R & D, developing digital communication systems, and teaching analog/digital electronics both at a technical college level and in industry.

Joseph DeMarinis (Book Review: *Current-Driving of Loudspeakers*, p. 18) is a retired Electrical Engineer with about 50 years design and engineering management experience in the fields of Television, Audio, Computers, and Digital Imaging. Now living in the greater Boston area, he has established a multi-purpose laboratory (www.WinchesterLab.com) with emphasis on microphone and loudspeaker calibration. He also has a busy schedule doing on-site concert recording for several Boston area orchestral, chamber, and early music groups. Born and educated in New York City, his active interest in DIY and professional audio goes back to the days when the term HiFi came into existence.

Tom Lyle (Review: Emotiva XPA-2 Power Amplifier, p. 24) has written for the web magazine Enjoy the Music.

VENDORS

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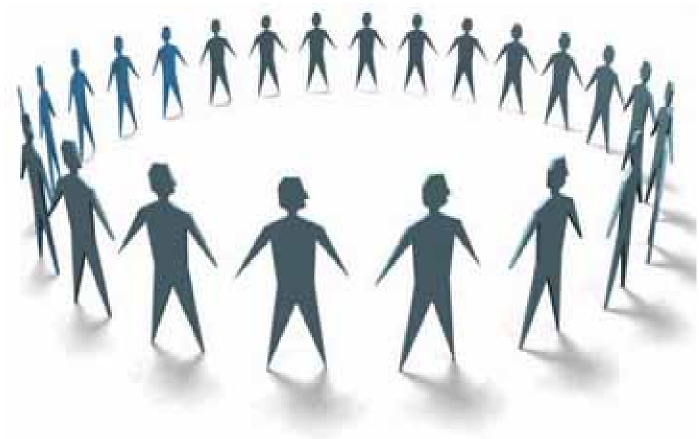
Ad Index

By David J. Weinberg (Tobias Engineering)

Future Technological Workforce



The Science, Technology, Engineering, and Mathematics (STEM) Education Coalition (www.STEMEdCoalition.org), co-chaired by the American Chemical Society and the National Science Teachers Association, comprises advocates from more than 1,000 diverse groups representing all sectors of the technological workforce who work to support STEM programs for teachers and students at the US Department of Education, the National Science Foundation, and other agencies.

"Companies have problems attracting and retaining engineers, and there is the thought that if engineering prospects are better understood, engineering work and organizations can be redesigned to more effectively appeal to them."

On Tuesday, 13 April 2010, representatives from seven of those organizations cosponsored a STEM-related presentation in the Rayburn House Office Building.

Congressman Vernon J. Ehlers (R, Michigan; and co-Chair of the House of Representatives' STEM education caucus) introduced featured lecturer Annalisa L. Weigel, MIT's Charles Stark Draper Career Development Assistant Professor of Aeronautics and Astronautics and Engineering Systems.

Companies have problems attracting and retaining engineers, and there is the thought that if engineering prospects are better understood, engineering work and organizations can be redesigned to more effectively appeal to them. There has been little beyond anecdotal data so far.

With that in mind, Dr. Weigel has been studying "Engineering Students, Their Motivations and Job Aspirations—How Can We Grow and Retain the Next Generation of our Engineering Workforce?" This was the first public presentation of her research results.

For this initial study she focused on aerospace engineering students who were sophomores or seniors during Spring 2009, partially because it is her field of interest and partly because it was easier to collect reliably representative data from this group as a basis for a longitudinal study that will cover the seven-year span from their sophomore year to five years after graduation.

These students were born during the last two decades of the 20th century (which anthropologists call the Millennial generation). Members of this age group are starting to enter the workforce and will comprise the new workers entering industry over the next 15 years. Thus, with the foundation of college data it will be easier to retain contact with them and collect reliable early-job-experience data that can be correlated with their college information.

Other reasons Weigel started with that group is her recognition that previous graduates tend to be hard to locate and contact, partly because they relocate often and to diverse sites around the world, and partly because there are substantial questions regarding recall accuracy when asking about their college experiences. These factors make trustworthy data collection more difficult.

This category of engineering students differs from previous generations in that they generally were nurtured children, readily multitask, assume the ubiquitous connectivity to technology, expect to change jobs throughout their careers and move often, want to excel and move up quickly in their profession, and don't consider rules as inviolate or institutions as relevant to their lives and jobs. These students have always had technology, the space shuttle, computers, and the Internet as part of their lives.

Weigel and her associates spent 18 months generating a 30-minute online survey for aerospace engineering sophomores and seniors, to assess their perceptions about their education and the industry plus their professional and job motivations and aspirations.

She hopes to follow up with the same group two and five years after graduation, creating a longitudinal data set spanning their college through early career years, plus providing feedback on career/work-related adjustments made by the colleges and the companies.

The survey asked about pre-college experiences regarding what got them interested in aerospace engineering, including what types of activities they participated in and if there were family or friends working in aerospace.

Investigating their college education, the survey asked how their schooling contributed to, or detracted from, their skills in 18 professional areas, as well as whether each of 11 other college-related factors influenced their decision to work in aerospace.

The survey pursued the students' expectations related to how long they would stay in their first job, whether they want or expect to work for one or more companies throughout their career, how often they might like to change projects, whether they want to move up to management, if they plan to switch careers, as well as their anticipation for escalating responsibilities, promotions and salary increases within their career field.

The survey (http://web.MIT.edu/caspar/aerosurvey.htm) received responses from about 600 students at 23 schools.

The students ranked salary, job excitement, job location, and work/life balance as the four most important job attributes out of a list of 16. Comparing aerospace with other industries, the students felt salary and excitement were the most positive, with location and work/life balance receiving about equal rankings for and against.

Of the respondents, more than half considered jobs outside their major field of study, and almost 60% wanted to pursue graduate school.

There is much more data analysis required, including gender and cultural preferences.

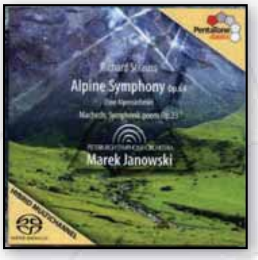
In discussions with her students, Dr. Weigel learned that they had a relatively negative reaction to job fairs, where they got to talk with many human resource people who were unable to explain prospective specific job duties and responsibilities; the students did not get to talk with engineers and engineering managers with whom they would likely be working.

The goal of her research is to make engineering students and professionals more productive and effective—to help them better meet their career objectives and desires, which will increase the likelihood that they will stay innovative and involved in engineering.

Companies in all technology fields might want to consider funding this type of research because it relates to their employees and work structures. E-mail AeroSurvey@MIT.edu for more information. aX

Super Fidelity

By John Sunier | www.audaud.com



Strauss | **Alpine Symphony; Macbeth** | Pittsburgh Symp Orch | PentaTone MC SACD PTC 5186 339 [Naxos]

Strauss's Alpine Symphony is a symphonic poem, not a symphony. It portrays a 24-hour experience of climbing a mountain and uses one of the largest number of players of any of his works—about 125.

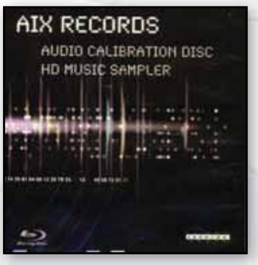
The work may be familiar to some collectors from the very first DGG CD which was pressed in 1983—a performance by Herbert Karajan still considered one of the milestone recordings of the work. The writing is some of Strauss' most descriptive, and some of his ideas have been lifted by soundtrack composers ever since. His tone poem Macbeth opens this disc.

It's exciting. The lower orchestral instruments portray Macbeth's theme, and Lady Macbeth's theme is heard in the woodwinds. The moods vary between heroic, warlike, and anxiety about terrible things. A glorious recording.

Art of Saxophone | **Glazunov, Debussy, Ibert, Milhaud, etc.** | Mario Marzi | Arts MC SACD 47748-8, [Albany]

This is an excellent collection of works for classical saxophone and symphony orchestra, superbly performed and recorded with state-of-the-art hi-res surround. Mario Marzi teaches saxophone at the Verdi Conservatory in Milan and is one of the world's leading classical saxophone virtuosos. The Glazunov sax concerto was composed in 1934 and has been much recorded, but this is one of its best performances and the remainder of the program is equally well-chosen and played. There is a preponderance of French composers but since the French invented the instrument. . .

While the sprightly Scaramouch Suite is fun on two pianos or with sax accompanied by piano, it really shines in this version with full orchestra. Sonics are first rate and for the technical mavens the inside of the note booklet cover has complete technical information.



AIX Records | **Audio Calibration Disc & HD Music Sampler** | Blu-ray (2009)

There are many calibration discs on SACD and DVD-A, but this one from AIX is one of the very first for use on Blu-ray players. I found the tests for proper phase between the various speakers of a multichannel setup, as well as the subwoofer tests, to be easier to use than any of the other test discs I have sampled. One problem is that there is only a channel ID test for 7.1-channel DTS-HD and none for 5.1—the prevailing codec for most Blu-ray movie discs. AIX has released some fine classical and jazz performances in the past, but for this sampler they only chose a single classical selection. I didn't find the other varied selections especially to my taste; however, there are fine examples of state-of-the-art screen images and hi-res surround sound.

Ben Webster | **My Romance (Audiophile version)** | UD-SACD 8934.2 [www.topmusic.com]

The sessions that produced these dozen tracks were recorded in Copenhagen in 1965, with Webster using mostly local musicians. Ben Webster was one of many black American jazz men who moved to Europe in the 1960s after being a mainstay of Norman Granz' Jazz at the Philharmonic concerts, and playing with Ellington.

Nobody else achieved quite the rich, warm, soulful, and caressing tenor sax sound so distinctively laid down by Ben Webster. And I don't believe I've ever heard it so cleanly and realistically as on this reissue disc. This is great stuff. The CD layer doesn't sound seriously compromised either. Track List: Brother John's Blues, Nancy with the Laughing Face, Duke's in Bed, What's I'm Bed, My Romance, Days of Wine and Roses, Blue Light, Stompy Jones, I Can't Get Started, The Theme, Friskin' the Frog, and Stormy Weather.

Handel | **Organ Concertos, Chaconnes & Fugues** | Richard Egarr | Harmonia Mundi SACD 807446.48

This is a set of Handel's last instrumental concertos, published after his death. In addition to the six organ concertos of Opus 7, Richard Egarr has included the "Cuckoo and Nightingale" Concerto.

He switches from being the organ soloist on a chamber pipe organ to the harpsichord for three Handel keyboard works that also date from the last period of the composer's life: A Chaconne and Fugue, and the Chaconne in G Major. The latter is a serious demonstration piece that uses exactly the same bass and harmonic pattern used by Bach in his Goldberg Variations.

The closest hi-res competition is the set of four SACDs in PentaTone's RQR 4.0 series. The Egarr version has a larger and richer-sounding chamber orchestra, but the PentaTone a better organ sound.

Grieg | **Percy Grainger & Edvard Grieg piano rolls** | Blu-ray & SACD 2L

My first reaction on seeing the latest in the courageous series of music-only Blu-ray + SACD sets from 2L in Norway was surprise at their choosing to release yet another version of the most popular piano concerto on earth. Then I saw the credit for the Grainger and Grieg Duo-Art piano rolls used in this performance and I understood.

There is a hint of the mechanical in the piano playing, but in general this is a fascinating re-performance—particularly since 2L has recorded the orchestra in a radial setup around the conductor and listener's sweet spot. The piano is front and center, but some of the first violins and all of the seconds are off to the left and right sides, with violas and timpani in the rear. I heard no difference between the SACD and either of the 5.1 or 7.1 Blu-ray lossless options.

audio on the avenue

By David Rich

The author, a long-time reviewer and commentator on the audio scene, will appear from time to time in this space.

WHERE DID TAPE-OUT JACKS GO?

In the recently introduced line of Denon audio/video receivers that add new features throughout the line including interfaces for 3D and iPods, something had to give to keep the prices constant. For Denon (www.denon.com), reducing the number and types of inputs and outputs on the back was the relief valve. With HDMI becoming a standard, it would not be a big surprise to see some S-video or component inputs disappear on low-end models...

READ MORE

HOW MANY SACDS DID THEY SELL LAST YEAR?

The rather depressing *New York Times* article, "In Mobile Age, Sound Quality Steps Back," published May 9, 2010, is required reading. It is best to read the piece in its entirety, rather than just my summary (www.nytimes.com/2010/05/10/business/media/10media.html). My focus here are the comments related to DVD-A and SACD:

"In 2003, 1.7 million DVD Audio and SACD titles were shipped, according to the Recording Industry Association of America. But by 2009, only 200,000 SACD and DVD Audio titles were shipped"...

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THIS CAT STILL HAS SOME LIVES LEFT

From the 3.3 in 1993 to the 2005-vintage Xd DSP active system, NHT has been responsible for many innovative products regarded as best-in-class despite their rational price tags. Unfortunately, NHT (www.nhthifi.com) has been bought and sold at least three times, with many ups and downs as a result. The last owners sold the company back to founder Chris Byrne but, soon after in February 2009, he announced NHT was "going quiet" with major layoffs and inventory sales. Well, the cat has survived again. On June 23, 2009, Chris announced the company was returning as an "entirely web-based entity."

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NHT's Absolute Tower speakers.

CALL US PROFESSIONAL

The restructuring of Harman International has been an ongoing process since Sidney Harman stepped down as chairman. On April 29, 2010, Harman decided it was time to let investors and analysts take a detailed view of the company in an all-day strategy update. I did not attend, but the PPTs and webcast are available on the Harman site (www.harman.com). Presentations were given for the consumer, automotive, and professional-products divisions. The big surprise was the composition of the professional division...

READ MORE

By Barry Fox

MS on Audio, Demos New & Old, Wall Wart Madness

This year brings a new threat and new opportunity for the hi-fi industry.

For the last ten years computer companies Microsoft and Apple have been banging out the message that a home computer "Media Center" makes the ideal way to listen to music, tune into radio, watch TV, and record broadcasts.

And for the last ten years the world has been saying "no it's not" and continuing to buy stand-alone audio and video and TV boxes.

Now there is a new campaign, fueled by a downturn in the sales of home computers. Most people who can afford a computer for work and Internet surfing already have several. This hits Microsoft harder than Apple, because Apple is creaming huge profits from iPods and iPhones, while Microsoft's Zune portable player has failed miserably.

Microsoft will now try even harder to convince us that a home computer makes a perfect replacement for audio and video stand-alone components.

Although modern hi-fi components are computers dressed up to look like old fashioned amplifiers, tuners, and recorders, there is one hugely important difference that gets lost in the hype.

Home computers are controlled by operating system software that can do a thousand and one different things and continually needs changing to let it do more things. Because this software is designed to be changed and stored on a magnetic disc it can equally easily be hijacked by the viruses and "malware" that circulate on the Internet or hide in music or picture files that people share.

So a home computer needs additional software to protect it against the malware. This also needs continual update changes to counter the dozens of new threats that crop up daily.

A stand-alone audio or video component is designed to do one thing only, and do it well. So it can be controlled by operating system software (such as Linux) that is frozen into microchips. It cannot be easily changed so cannot be infected from the Internet or recordings.

Also, the stand-alone component starts up quickly because the control signals are coming at very high speed from a chip, not slowly from a magnetic disc. Switching off or being shut down by power failure cannot do any damage to the control software because it is frozen into the chip.

Why remind on this now? Because Microsoft recently embarked on a new and major campaign to convince us that the latest version of Windows (called 7) finally does what all previous versions of Windows have promised to do but failed to deliver, and makes the perfect home entertainment tool or toy.

Don't believe it. I have installed Windows 7 on several PCs and although it is less likely to crash than previous versions of Windows, it has done nothing to change my conviction that a home computer is a rotten replacement for stand-alone hi-fi components. Apart from anything else, stand-alone hi-fi plays music while a home computer is out of action doing yet another update.

Things that worked with previous versions of Windows no longer work or don't completely work with Windows 7. Microsoft has perversely changed the look and feel of commands, so actions that have become second nature now must be relearned. It's like buying a new car and finding some of the controls are in different places, or missing until installed as optional extras.

If you are buying a new computer then it will come with Windows 7, and you will at least start with a working machine. Upgrading from the previous and hugely unpopular Windows Vista should be reasonably easy. But upgrading from the previous and much more popular Windows XP can be a painful experience. I know because I have done it. It's a classic case of breaking something that wasn't broken by trying to fix it.

So Windows 7, like all previous versions, makes a wonderful advertisement for all those companies selling real hi-fi.

DEMOS NEW & OLD

There is a whole generation now of new music-lovers that has only ever listened to music in MP3 and iTunes

AAC, played from a PC or portable player, through tinny earbuds or in a cinema.

At a recent exhibition Sennheiser was demonstrating high quality headphones with horrid MP3 pop music played from a laptop. When we asked for better source material the young demonstrator just looked blank.

So spare a thought for Gilbert Briggs, the Brit who founded Wharfedale, and in 1955 booked the Royal Festival Hall in London to stage the ultimate comparison, a concert that mixed live and recorded music.

The idea came from a small event Briggs had arranged in Bradford the year before. At the Bradford concert a pianist mimed to the sound of a piano recording reproduced through a Wharfedale corner cabinet. When the audience realized that what they were hearing was a recording rather than the real thing, they applauded.

"You're mad," his friends and staff said when Briggs told them he planned a bigger and better do in London's then-new Southbank Centre. Even the Royal Festival Hall's manager tried to stop Briggs wasting his money. When the tickets were printed, the RFH box office sent Briggs all but 250 of the nearly 3,000 available.

"We think 250 will be more than enough for us," they told Briggs snootily, "because we can't see who on earth will want to pay good money to listen to recorded music at a concert."

Nothing daunted, Gilbert Briggs advertised his concert in the musical and hi-fi press of the time and offered tickets by mail order. On the morning that tickets officially went on sale from the Festival Hall booking office, there was a long queue. By eleven o'clock the same manager was on the phone to Briggs in Yorkshire, asking for as many of the other 2,750 tickets as he could spare.

The concert was a sell-out. So were the three similar concerts that followed. After reminding the RFH that royalty had not shown any interest in attending, Briggs's family sat in the Royal Box. Peter Walker and John Collinson of Quad sat in another box operating the disc and tape equipment. On stage, Briggs had four corner cabinets, each driven by a 15W Quad valve amplifier.

There was no question of involving the audience scientifically or technically. The concert was a concert, but with a difference, because live musicians played together and in alternation with commercial recordings on disc and specially recorded tapes. EMI, Decca, and the BBC all cooperated with the recorded program material, which was both in mono and stereo; a collector lent his own high quality sound effects recordings, such as the sound of a tugboat.

When I met Briggs toward the end of his life he told how the only real problem arose during the tugboat sequence. The recorded effect was on disc, and the turntable spewed its ball bearings halfway through the effect. The tug ground to a halt. But Briggs had arranged for every piece of equipment to be backed by a spare with duplicate recording.

The London Mozart Players under conductor Harry Blech were Denis Matthews (piano), Campoli (violin), Leon Goossens (oboe), and Ralph Downes (organ), all played live. One concert ended with the Hallelujah chorus; the LPO and choir conducted by Sir Adrian Boult appeared on disc through the corner horns, with Ralph Downes at the organ and the London Mozart Players on stage.

Two sell-out events at Carnegie Hall followed with RCA providing the amplification and tapes, all for no charge. But, as Yorkshireman Briggs recalled sourly, "The electricians were charging us £25 a time just to switch on a switch."

During the hi-fi boom years of the 1970s there was talk of a repeat performance in London, but Briggs died and Wharfedale changed course.

I wonder what would happen if someone tried it today?

WALL-WART MADNESS

Strange but true. People who need 96kHz and 24 bits through valve amps and studio monitors at home will cheerfully accept far lower fidelity for music on the move from battery portables. But no one can be cheery about the insane lack of a standard for portable chargers.

Manufacturers build to different voltages, polarities, and power drains. Often there is back-voltage sensing to prevent over-charging, because an over-cooked lithium ion battery can easily explode and burn fiercely. So the makers fit different connectors to stop people using the wrong chargers, or build intelligence into the charger so that only approved batteries will work with it.

On a recent trip to the US my bag of music, phone, and camera chargers and leads weighed more than my clothes; and I still destroyed an expensive piece of kit because I tried to use the wrong charger.

There is a move toward 5V as a de facto standard, because that's the voltage that feeds through a computer USB connection. Mains wall and car power units now often deliver 5V to a portable through a standard USB lead to either a Mini or a Micro USB connector.

But small USB sockets are fragile and if you break one the whole device may be a write-off. Some USB chargers will refuse to work with some chargers. This may be simply because the charger is not delivering enough current or it may be because the charger and the portable do an electrical handshake before charging starts. USB uses four wires; two for the 5V supply and two for computer data.

It is possible to short the USB data wires to fool the intelligence, but there may then be a risk of damaging the portable.

If you believe a recent spate of publicity, the answer to our prayers lies in inductive charging. A flat pad with charging coils inside plugs into the mains to act as a transformer primary, and different portables sit on the mat behaving as transformer secondaries to pick up a wire-free charge.

"Lose the cords—unplug" promises UK/Israeli company Powermat in its advertising blitz. Rival products are on the way, for instance from Duracell.

There is nothing new or magic about inductive charging. It's how electric toothbrushes work. But the brush is matched to the mat. Powermat promises to charge three different devices at a time, plus a fourth from a USB port on the mat.

If only it were that simple.

The portable must be fitted with a "receiver"; for a Blackberry it's a new back, but for iPods and iPhones it must be a slide-on sleeve. This increases the cost; £70 for the mat (£80 if the mat folds for travel) and its wall power unit, and £30-£35 for each receiver.

Ran Poliakine, Chairman and CEO, told me: "When the device is fully charged the mat goes into standby and stops drawing power, whereas ordinary wall chargers continue to waste power. And with three devices there is less power drawn than with three wall chargers."

I used a digital power meter to compare drain when charging an iPod Touch with a Powermat against a simple USB wall charger (a Camelion, bought from Maplin, the company that filled the gap left by Tandy Radio Shack's loss of interest in Europe).

The supplied receiver sleeve would not fit my Touch because my Touch is a first generation model and Powermat only sells sleeves for second generation Touch iPods, which are slimmer. Yes, Powermat's packaging is clearly marked "iPod Touch 2nd generation" but customers with 1st and 3rd generation Touches need to know to check before buying.

I then tried charging the Touch from the USB socket on the mat. This would not work, because iPods need to do a data handshake with the charger. This came as news to Powermat's UK technical support.

One option is to use Powermat's £30 power "cube" which sits on the mat and charges mobiles by cable through a selection of eight tips. But this is hardly in the spirit of wireless charging. Another option is to use an inductive iPod dock which has a standard iPod connector. But this seems an expensive alternative to using an ordinary iPod dock.

My power tests showed that charging a single device with the Powermat offers little or no power saving, or convenience benefit, over a wall charger that draws current only when charging. Charging two or three devices at the same time will clearly be more convenient and may save a little power—but all the adaptors needed will cost a packet.

Ran Poliakine admits: "This is only the beginning. Our next step is to build the receiver into a replacement battery. By January 2010 we will have 15 batteries with built-in receivers."

Some designs were shown at CES, but none is yet to be seen in stores. So we are still waiting for the promised lost cords. *aX*

CONTRIBUTORS

David J. Weinberg ("Future Technological Workforce") is an engineering consultant and technical journalist on audio, video, and film technology.

John Sunier ("Super Fidelity") is a CD reviewer for *Australian HiFi and Home Theatre Technology*. His website is www.audaud.com.

David Rich (Audio on The Avenue) received his MSEE from Columbia University and his Ph.D. from Polytechnic University of NYU. He specializes in the design of analog and mixed-signal integrated circuits and has taught graduate and undergraduate courses in integrated electronics and electroacoustics. Student work under his guidance, including a novel high-efficiency mixed-signal integrated power amplifier, has won numerous awards. His industrial positions include Technical Manager at Bell Laboratories. His portfolio has spanned the design of audio ICs for Air Force One to RF ICs for wireless cell phones, and his innovations have earned 14 patents. He is a Senior Member of the IEEE and has frequently served as chairperson for technical and panel sessions at IEEE conferences. He has been a member of the AES signal processing technical committee and has been Technical Editor for *Audio Critic*. He is the head of the music committee of the Bethlehem Chamber Music Society.

Barry Fox ("Media Report") reports on the audio industry as columnist for the British publication *Hi-Fi News*.

CORRECTION

The following contact information was omitted from the recent review of the Bryston BCD-1 CD player (June *aX*, p. 22):

Bryston Ltd. is a Canadian company with authorized dealers in countries worldwide as well as throughout the US and Canada. To find the dealer nearest you, go to bryston.com and click on "dealers."