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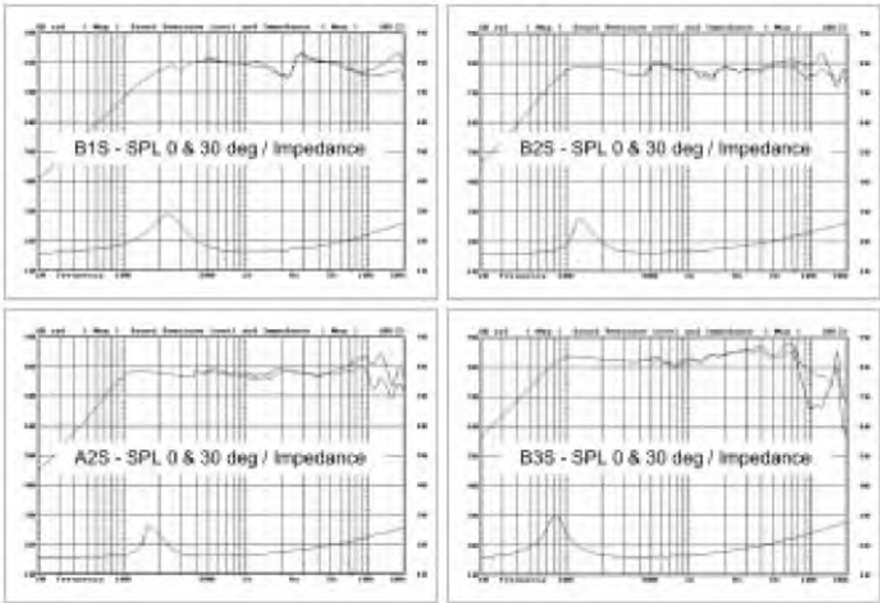

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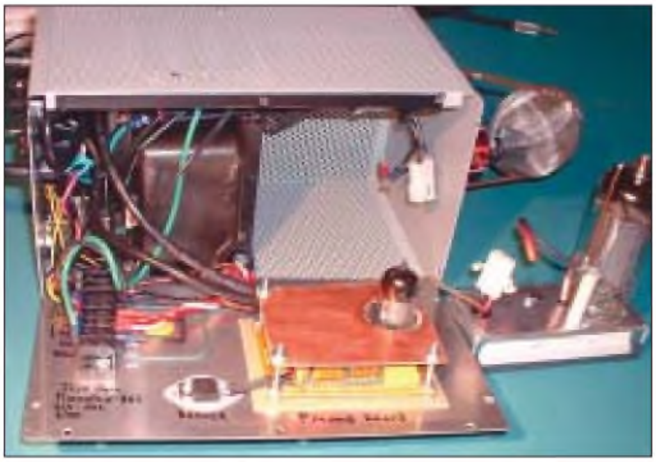

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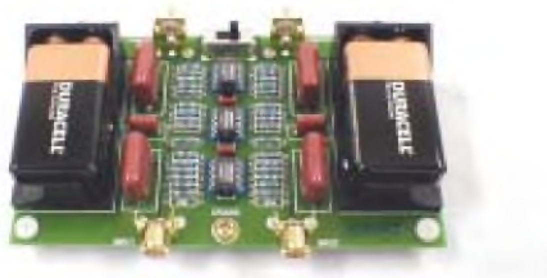

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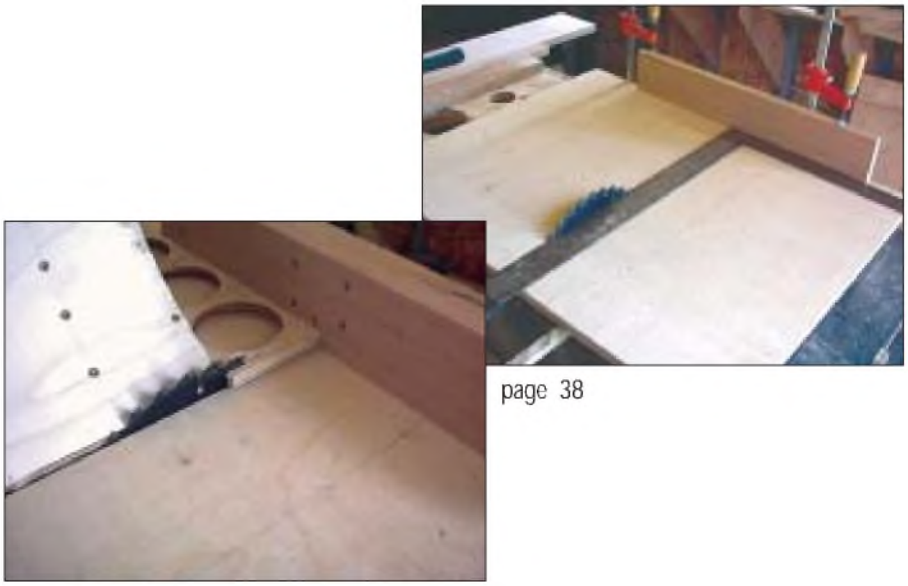

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solid state tubes speakers

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JOHN STUART MILL

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Build a Plasma Tweeter

Generate audio from a massless electric flame to produce perfect transient response and an omnidirectional frequency response. **By Colin Joye**

PHOTO 1: The finished project.

All dynamic cone, electrostatic, and ribbon tweeters have a small mass associated with their respective sound-producing elements. A tremendous amount of work has been done to minimize this moving mass because it is often the limiting factor of how well the tweeter will perform. Another drawback to loudspeakers that use moving elements to produce sound is directionality.

Any element that moves to cause sound does so by pushing the airwaves in a specific direction. Some tweeters combat this by using a horn, others by using multiple drivers. In the case of the former, the frequency response is not perfectly uniform over the angle of dispersion, while in the latter, a lobing effect is often unavoidable. Having a uniform frequency response for off-axis angles prevents the listener from being confined to a single "sweet spot," where all of the axes of all of the speakers converge.

Of course, if your ears are not equidistant from any pair of sound sources, you will experience a phase shift associated with the different path lengths, but you still want the same frequency response, even off-axis. This project doesn't constrain you to experience only one fixed soundstage, but allows the soundstage to rotate before your ears as you walk around the speakers, adding to the illusion of "being there."

ABOUT THE AUTHOR

Colin Joye is a 2002 graduate of Villanova University in Electrical Engineering. He is currently pursuing his PhD in electromagnetics at Massachusetts Institute of Technology. His other interests include music composition, art, and karate.

ION TWEETER SOLUTION

You can use the laws of diffraction to make the width of the dispersion field wider by decreasing the surface area of the moving element with respect to the wavelength of sound being emitted (*Fig. 1*). The problem is that, by reducing the surface area, the sound output level is also reduced and the frequency response becomes restricted, so this would be a rather impractical solution to the problem. One clever way to conquer both directionality and moving mass is to use an electric flame.

Dr. Siegfried Klein invented the ion tweeter (or plasma tweeter, as it is also known) in the early 1950s. He used a push-pull vacuum tube oscillator to drive a coil at its self-resonant frequency to cause voltages on the order of 30,000V (30kV), which he used to produce corona discharge.

One problem that was prevalent in power transmission lines in the 1920s and 1930s was the formation of corona discharge on power lines. When high voltages are present at sharp points or edges, a bluish glow discharge known as corona discharge forms around the sharp point. Corona discharge is a localized ionization of the air that conducts electricity. Any such gas that conducts electricity is known as plasma.

In power transmission lines, this corona caused the insulation to disintegrate, because corona is a hot glow discharge. Dr. Klein's invention made use of this corona to produce audible sound. He touted it as being able to produce sound with tremendous efficiency (he was ignoring the oscillator itself, which actually draws a lot of power) and as being able to project sound very audibly over large distances (as described in U.S. patent number 2,768,246).

In the late 1950s and 1960s, ion tweeters were produced commercially in the USA and Europe under the names Ionovac, Ionofane, and Ionophone (I recently spotted one for sale on eBay.com). These units used a metal horn to focus the sound and facilitate mounting. They used a single 6DQ8 tube to produce the corona and used tubes in the amplification stages.

Since then, there have been few ion tweeters produced commercially. The German company Magnat produced a three-way speaker cabinet in the 1980s and used an ion tweeter without the horn to make use of the omnidirectional effect. Acapella, another German company, currently produces very expensive and very high quality speakers that use ion tweeters.

This project presents the ion tweeter that I have constructed (*Photo 1*) and illustrates its theory of operation, construction techniques, feasibility, and some of the observations I've been able to make. I actually got the idea for my first ion tweeter on the web from Ulrich Haumann's webpage, now http://www.plasmatweeter.de. After I read how it worked and what it did, I just had to build it.


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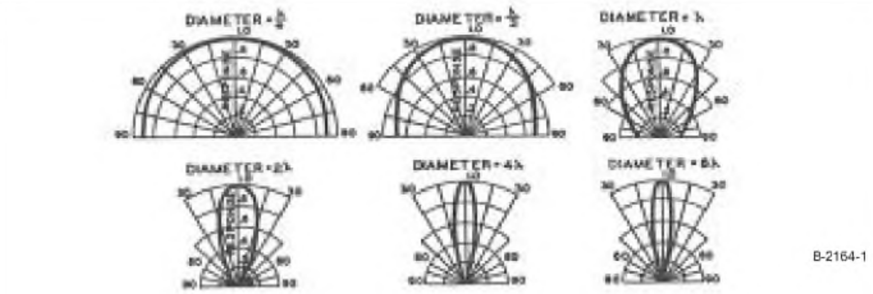


FIGURE 1: Normalized directionality characteristics of a circular piston source as a function of diameter and wavelength (Ref: Olson, Harry F., *Music, Physics and Engineering*. New York, 1967, 2nd edition, p. 103).

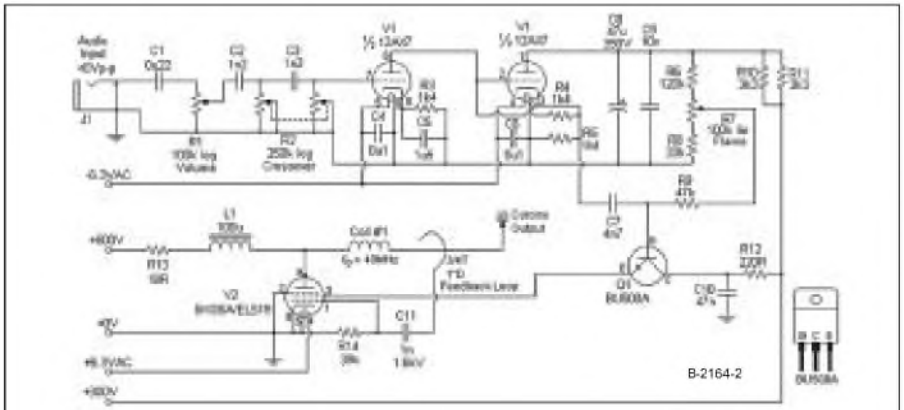


FIGURE 2: The Plasmatone-RG1 preamp and oscillator schematic. The preamp section is the upper half of the schematic, while the oscillator section is the lower half.

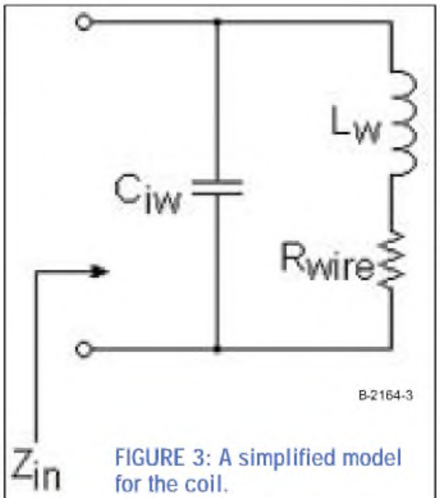


FIGURE 3: A simplified model for the coil.

The ion tweeter schematics I used for this project are similar to the first pair I built, with a few modifications, such as the 12AX7 tube preamp stage. The first pair has a lot of problems, so I built a better, more reliable second pair, as presented here. I am presenting the schematics exactly as I have used them, but there are many things I think can be enhanced in the schematics, so I have included a section of recommendations that you should read before attempting to build this project.

ABOUT THE FLAME

The corona flame is very peculiar because it stands by itself on top of a single electrode, not between two electrodes, like a Jacob's ladder or spark-gap transmitter. This feat is quite mind-boggling to the typical experimenter who is used to seeing sparks appear between two points. In fact, even the resonant coil that is the heart of the project is only connected at one end! The ion flame looks a lot like a candle flame (*Photo 2*), although it is slightly bluish or white in color. Furthermore, the flame acts like a perfect point source, which means that the acoustical energy is emitted spherically with a perfect uniformity in frequency response.

I like the ion flame because it operates on the same principle that nature uses to produce sound by lightning and thunder. Unfortunately, the low-frequency extension of the flame is limited by its size. You probably could have guessed that, since thunder can easily reach down to 1Hz or less, requiring millions of watts of electricity in the process, while the spark you get stepping out of your car on a winter day gives you just a little more than a high-pitched snap.

For a 1cm flame height (around the same size and shape as a medium candle flame), the frequency response hits –3dB around 2kHz. The upper limit of the frequency response is well above the hearing threshold. It has been measured up to 40kHz, but my guess is that it may extend up into the megahertz. If you were to send bass frequencies to the flame, it would be like sending bass frequencies to a tweeter. It will try to reproduce the frequencies, but it won't do it very well!

This project draws around 200W of wall power to produce that 1cm flame, so if you were thinking of making an ion *subwoofer* next, forget it! You would need a lot of power. Subwoofers tend to be non-directional in small rooms anyway, so only the higher frequencies really need to be addressed.

Those of you familiar with spark generation know that where there are sparks, there is the pungent smell of ozone. The corona flame, however, is not a spark, so there is almost no ozone. Since the flame is hot, the ozone that it does produce rises straight above the flame, but there really isn't enough of a smell to cause an annoyance or health hazard. Besides, ozone kills bacteria in the air!

The corona flame does generate a lot of heat, and it is a high-voltage hazard, so be very careful when you are working around it. Furthermore, it acts like a little source of a lot of radio frequency interference (RFI). The Federal Communications Commission (FCC) has a lot to say about such things, so I bought a steel mesh food strainer to put around the flame (*Photo 3*).

This grounded shield allows the air-

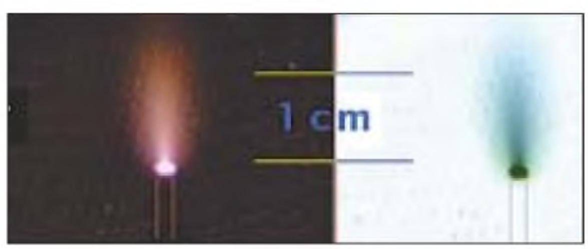
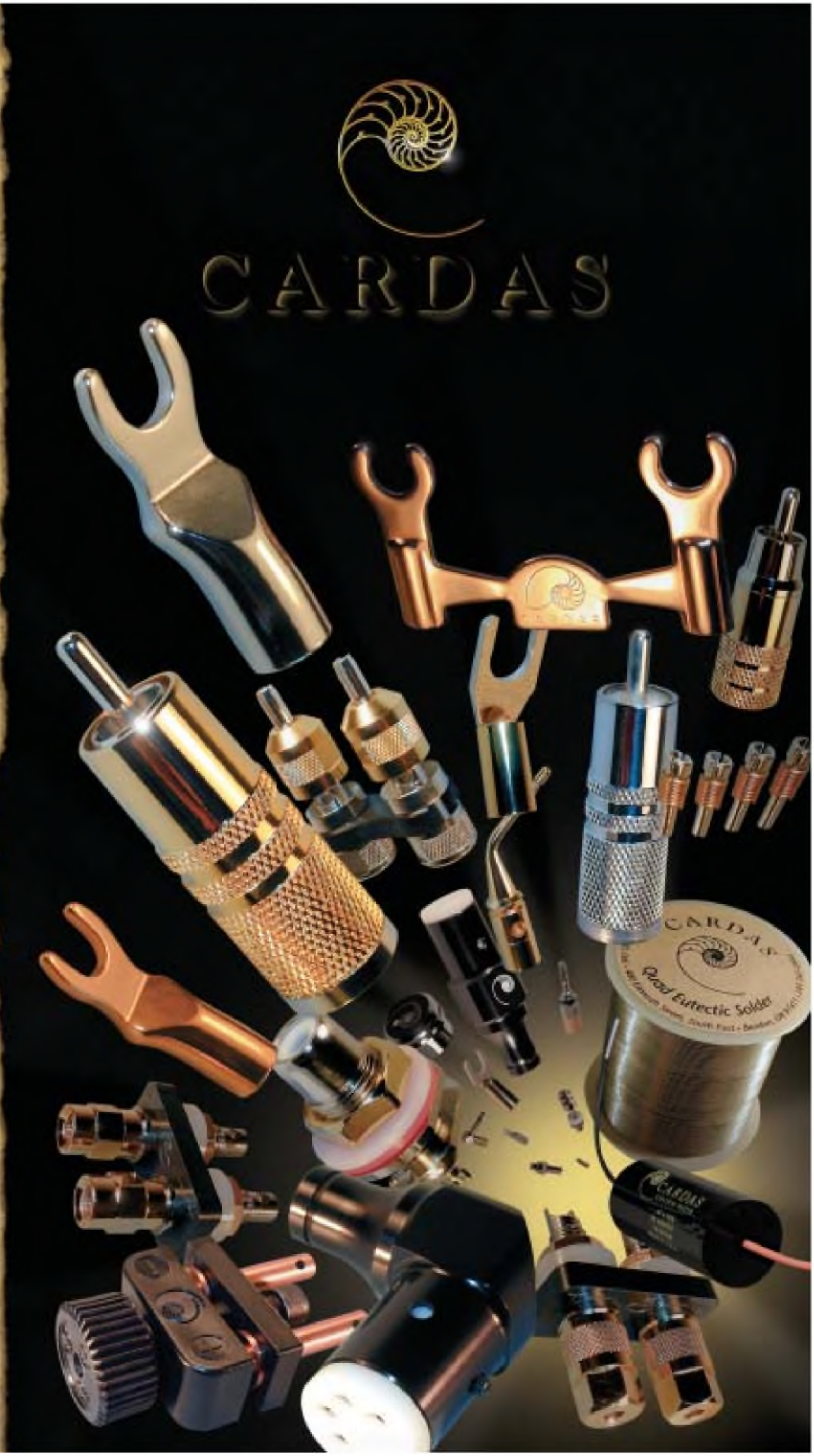


PHOTO 2: The corona discharge on the electrode (left); shown with inverted colors (right).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waves to pass through, but prevents electromagnetic waves from propagating. You can see how it works if you pass a small fluorescent tube by the flame without the shield—it will light up 6″ from the flame! Incidentally, since the oscillator frequency is very high (around 30MHz), you won't get a shock from the flame, but a radio frequency (RF) burn. Since the flame is so hot, you need to use a copper electrode to sink the heat.

I tried a piece of stainless-steel wire once, but it melted! The copper still is slowly "eaten" away by the flame, but there aren't many other options for the electrode material. In the 1960s, they used thoriated tungsten quartz-glass electrodes that retained their sharpness for up to 1200 hours.

PHOTO 3: The Faraday shield is simply a food strainer.

CIRCUIT OPERATION

Figure 2 shows the schematic I used for the preamp, amp, and oscillator portion of the project. I like to start with the oscillator, because explaining it first helps explain other parts of the system.

The heart of the oscillator is what I call the "Tesla" coil. It is not the Tesla coil some of you may be familiar with because it doesn't have a primary and secondary, but it does operate at the self-resonant frequency of the coil. This coil acts similar to a parallel resonant circuit (*Fig. 3*), where the windings are the inductance component and the interwinding capacitance leads to the capacitive component. The key to producing the high voltages is to have a very high Q-factor for this resonant circuit. By definition,

$$Q = (2\pi f_0)/\text{bandwidth} = (2\pi f_0 L)/R$$

where f_0 is the resonant frequency in hertz, L is the inductance in henries, and R is the resistance in ohms.

So to get a high Q, the resistive losses need to be small and the inductance large. Since the coil acts like a parallel resonant circuit and the losses will be small, then at the self-resonant frequency of the coil, the coil looks like a near-open circuit (*Fig. 4*). From Ohm's law, if a current is put across a very large resistance, the result is a very high voltage.

In his 1956 U.S. patent number 2,768,246, Dr. Klein noted that an audible hiss was present if the oscillator operated at frequencies below around 3MHz, which sets the minimum acceptable self-resonant frequency. Actually, I found that the resonant frequency should be kept above 10MHz to avoid flickering of the flame. The 6KG6A pentode I used for the oscillator has a gain-bandwidth limit of around 70MHz for this design, so I wanted a self-resonant frequency of around 40MHz for the operation. The reason I chose it so high was that a lower resonant frequency would require more inductance, which means more wire and more losses.

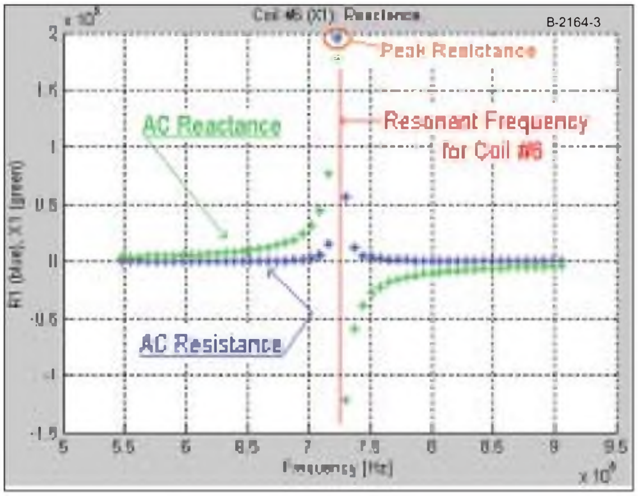


FIGURE 4: An impedance plot for a typical coil near self-resonance.

In order to design the coil to oscillate at a certain frequency for a parallel resonant circuit with negligible losses, you can use the following simple equation

$$f_0 = [2\pi\sqrt{(C_{iw}L_w)}]^{-1}$$

where C_{iw} is the interwinding capacitance and L_w is the winding inductance. Unfortunately, there is no practical way to accurately estimate C_{iw}, so I wound several coils until finding a suitable one.

By the way, you cannot simply make the inductance extra small to lower losses and then put a high-voltage capacitor in parallel with it to lower the resonant frequency, because the Q-equation would predict a small Q. Also, you can't use Litz-style wire (a bundle of small-gauge wires to lower loss) because corona favors sharp edges! The only options left are heavy-gauge copper magnet wire or superconductors! I chose 16ga copper magnet wire because it seemed to do the job well without too much loss or being too bulky.


PHOTO 4: Resonant coil, little feedback loop, and Faraday shield.

My coil is wound with a diameter of 1⅜″ (35mm) with 15 turns. Since I didn't have a ceramic, glass, or otherwise suitable core to wind the coil on, I wound it on a cardboard tube and then threaded the turns together so they wouldn't move when I took the coil off

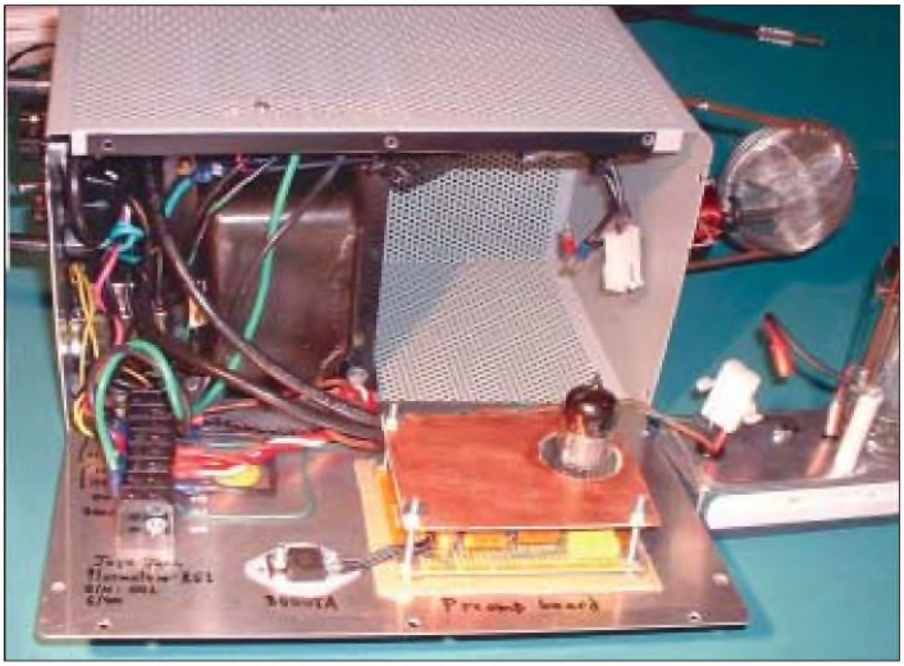


PHOTO 5: The disassembled ion tweeter.

the cardboard. Each turn is then separated by the width of a piece of thread.

The resonant frequency of these coils is around 40MHz, as desired. I don't think there would be any problems if the resonant frequency was as low as 10MHz or as high as 50MHz. I dared not wind the coil on anything that would melt at high temperatures after I found that the coils became quite hot on the first ion tweeter pair. I originally thought this heating was due to the resistive losses, but now I think it was simply heat conducted away from the hot corona like a heatsink. I think it would even be okay to wind the coil on PVC piping, provided there is no direct path for the heat generated by the corona to use your coil as a heatsink.

Place a simple feedback loop around the output electrode in order to tell the oscillator tube when to turn on and off. This feedback loop is nothing more than an insulated wire placed on the output-side of the resonant coil. It does not touch the output electrode, but hovers near it, no closer than about ¼″ (6mm). My feedback loop is around 1″ in diameter and is securely fastened to prevent it from moving too near the electrode. See *Photo 4* for the coil and feedback loop assembly.

INDUCTOR SELECTION

The 100µH inductor is simply an RF choke that lets DC pass through to the tube, but blocks AC from coming back to the power supply. I tried several different inductors and found that some kinds actually didn't work very well due to lossy ferrites with low Qs. The best

PHOTO 6: The simple oscillator circuitry is mounted under the 6KG6A socket.

PHOTO 7: The 6KG6A is mounted in the front of the enclosure.

choke was the Xicon 1A, 100μH inductor that was wound using only one layer (which prevents inter-layer coupling).

Curiously, when I put it on the impedance analyzer, its self-resonant frequency of around 3.2MHz was well below the operating frequency of the circuit (around 40MHz), which means it should be acting more like a capacitor at that point than an inductor, although probe capacitance may have thrown the resonant frequency measurement off. Nonetheless, it works, but I'm wondering whether this could be causing interference in my 120V power lines.

By the way, I did experience interference problems with my portable CD player. Since portable CD players are not normally shielded, they easily pick up interference from the air or power lines. My first ion tweeter prevented my portable CD player from functioning properly whenever the flame was on!

TUBE OPERATION

When you first turn on the power, the sudden disturbance causes the self-resonant coil to begin to oscillate. The feedback wire senses this and triggers the tube to shunt the power rail toward ground. When that happens, the current in the coil shifts and causes further disturbance. At this point, the circuit is oscillating, but no corona is present.

In order to initiate the corona, you must momentarily touch an isolated screwdriver to the tip of the electrode. The proximity of such a conductive body causes the voltage gradient (e.g., kilovolts per centimeter) to increase above the critical threshold of visual corona (Peek, F.W., *Dielectric Phenomena in High Voltage Engineering*, Third Edition, McGraw-Hill Book Company, Inc., 1929, pp. 54–59). The corona is self-sustaining once this critical threshold has been reached.

After constructing a solid-state version of this ion tweeter as a senior project, my team and I are just beginning to understand how to produce a self-starting corona. If the power to the oscillator is switched on quick enough, a burst of high-frequency energy content is produced at a substantial level. The high Q of the coil causes it to select the frequency it wants to see and reject all others. If this initial burst is at a high enough amplitude, the corona will self-start; otherwise, it will need to be started mechanically. My senior project team has been able to self-start the corona discharge reliably using this approach with solid-state devices (see http://www.ee.vill.edu/ion).

The 6KG6A pentode, like all pentodes, has three grids. Grid #1 is the oscillator grid because it is the most sensitive and has very low shunt capacitance. The low shunt capacitance is important because the feedback loop is nothing more than a wire placed near the electrode to sense the output by capacitive means. The shunt resistor between Grid #1 and ground acts as a self-bias to ensure that the tube sustains class C operation.

Grid #2 requires a DC bias to set the current flow in the tube, and thus the height of the flame. By adding an AC signal to this DC bias, the flame height can be changed very rapidly. The flame system works like a Class A circuit: the steady-state size of the flame is set by the flame control potentiometer, and the AC signal causes the flame to shrink down to a minimum size and expand up to a maximum size. If the AC component causes the corona to extinguish momentarily, the signal will be audibly clipped. Likewise, if the peak AC signal causes the tube to saturate, clipping will result.

If the steady-state flame height is set too high, the plate of the 6KG6A will quickly begin to glow orange, a sign that the tube is being operated outside of its safe area. The 6KG6A, like most power tubes, generates a tremendous amount of heat under normal operation.

WHY IT WORKS

One possible explanation for how the corona generates sound is that the flame is hot, so the air density is lower than that of the surrounding air. This causes an air density gradient boundary that moves with the flame height, producing sound waves.

Another explanation (as used by Dr. Klein) is that the bulk of the collisions between positive air ions and air molecules causes spherical piston-like variations in the air column, producing elastic pressure waves. There is no known upper limit to how high the sound pressure level could be made if the plasma flame was large enough. I have mea-

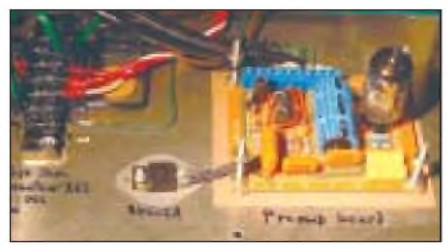


PHOTO 8: Preamp circuitry.

PHOTO 9: Power supply.

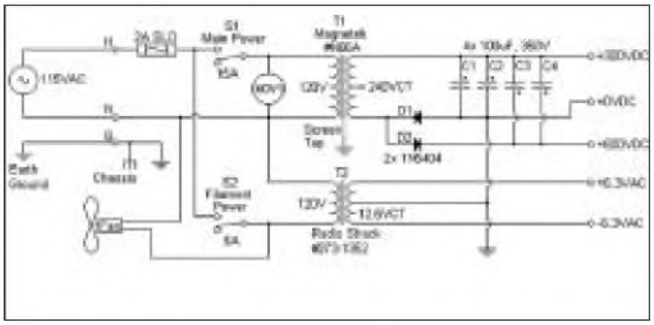


FIGURE 5: The Plasmatone-RG1 power supply.

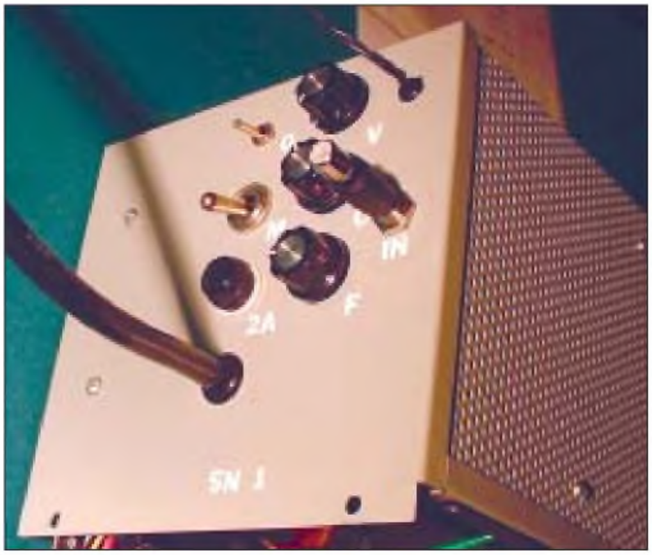


PHOTO 10: Ion tweeter controls.

sured sound pressure levels of 108dB at 3″ from the flame, but I'm sure it could become louder if the conditions were right.

Putting the audio and DC bias into the Grid #2 of the tube requires around 10mA at 20V to 100V (roughly). The maximum peak-to-peak voltage of the AC component should be around 50V. A simple BU508A NPN power transistor in common-collector configuration provides the necessary current gain to accomplish this interfacing of the preamp and oscillator tube. The BU508A is a horizontal sweep-type power transistor used in television, so it can handle a considerable amount of voltage and current. It really does not need a heatsink, but I mounted it on the aluminum cover just to be sure that it didn't overheat.

The DC bias level is set by the 100kΩ potentiometer (marked "Flame" in *Fig. 2*), while the 12AX7 tube provides the AC voltage. I found it was necessary to use the cathode-follower configuration for half of the 12AX7 because of the current needed to drive the BU508A power transistor. Because of this common cathode-to-cathode follower configuration, the overall gain is a bit lower than just one section of the 12AX7.

I didn't put any other gain stages in for some reason, but it really does need about 30dB more gain. To get full volume output with my schematic, you need about 6V of peak-to-peak audio on the input. Keep in mind that I am not much of a tube expert. I did no load line analysis for this preamp stage, and I'm not even sure whether you can use the term "small signal approximation" for a 12AX7 pushing 50V AC without nonlinearity problems.

The input of the preamp section filters the audio input by the simple variable high-pass filter, which has a variable cutoff point of around 500Hz and up. As mentioned earlier, it is important to have some kind of high-pass filtering somewhere along the road.

The power supply for the ion tweeters is shown in *Fig. 5*. I don't like it at all, because I used a voltage doubler circuit to obtain the necessary 600V DC. Such a circuit has increased 60Hz noise present and bad power-supply ripple. I suggest using a center-tapped (CT) transformer rated somewhere around 800V with a full-wave rectifier and choke-input instead.

I had problems with my power switches blowing out on the 600V/300V supply because of the large current spikes when turning the system on and off, so this is part of my motivation for recommending a choke before the capacitors. Also, I used a little 1.2A 12VCT transformer for the 6KG6A filament, which is rated at 2A, so you should obviously change that.

PERFORMANCE

Since I spent most of my time getting the concept to work, I really didn't do much in the way of tweaking it for perfect performance. I think there are some relatively severe nonlinearity problems somewhere in my design, but it's because I'm really only using it above 3kHz; it is not readily apparent to the untrained ear. I'm sure that proper designing and tweaking will address most of the nonlinearity, because I believe that the corona discharge is an excellent substitute for any kind of moving-element tweeter.

Everyone I've demonstrated it to thought the concept and sound quality were absolutely incredible. I really like the omnidirectional aspect of the tweeter, as well as the concept of massless sound generation. It is absolutely amazing to walk around the stereo system and have the high frequencies be perfectly uniform at all directions. It is perhaps even more impressive when the lights are turned off and you are greeted by the dancing white-purple electric flame and warm glow of vacuum tubes.

CONSTRUCTION AND TROUBLESHOOTING

I set out to build an integrated, sturdy ion tweeter when I began this project. To accomplish those goals, I put everything into a Hammond perforated-steel cage and simply drilled holes in the edge for the bottom plate.

In retrospect, I don't think it's a good idea to mount the power supply in the same case with the electronics, because I had a hard time getting rid of some extra noise. I also think they could look a lot more elegant if the tubes were showing and had some kind of Frank Lloyd Wright theme with tall skinny capacitors and tubes. I pretty much stuffed everything into the steel enclosure and then had no room for the fan I decided to add later to cool the 6KG6A tube.

Photo 5 shows the tweeter in its disassembled state. The 6KG6A tube and oscillator circuit (*Photos 6* and *7*) fit inside the front of the enclosure, while the preamp board is mounted on the bottom cover plate (*Photo 8*). The preamp board shown in *Photo 8* actually has some extra circuitry on it that is

WARNING!

This project utilizes potentially lethal voltages and should not be attempted by those unfamiliar with high voltages. The corona discharge is produced by voltages in excess of 20,000V and poses a tremendous heat and fire hazard if exposed to combustible materials. Furthermore, the operation of the corona discharge is such that it operates in the radio frequency range of 20MHz to 70MHz and is capable of causing radio frequency burns. This unit may emit severe radio frequency interference, so you should use the Faraday shield described in the text to block the RFI and protect the user from coming into contact with the corona discharge flame.

PARTS LIST

ION TWEETER PARTS USED, VACUUM TUBE VERSION

AUDIO PREAMP

REF	ITEM
C1	0.22µF, 250V capacitor
C2, C3	1.2nF, 100V capacitor
C4, C6	0.1µF, 63V disc capacitor
C5	1.5µF, 50V capacitor
C7	4.7nF, 100V capacitor
C8	47µF, 350V capacitor
C9	10nF, 400V capacitor
C10	47nF, 400V capacitor
J1	RCA jack
Q1	BU508A horiz power transistor
R1	100kΩ potentiometer, log
R2	250kΩ dual potentiometer, log
R3, R4	1.4kΩ, ¼W resistor
R5	1MΩ, ¼W resistor
R6	120kΩ, ½W resistor
R7	100kΩ potentiometer, lin
R8	33kΩ, ½W resistor
R9	47kΩ, ½W resistor
R10, R11	3.3kΩ, ½W resistor
R12	220Ω, ½W resistor
V1	12AX7 preamp tube
	PCB 4″ × 6″
	12AX7 tube socket
	RS #276-1388, PCB blue wire terminals (5)

HIGH VOLTAGE OSCILLATOR

REF	ITEM
Coil #1	16ga magnet wire (~6ft)
C11	1nF, 1.6kV disc capacitor
L1	Xicon 100µH, 1A RF choke
R13	18Ω, 10W resistor
R14	39kΩ, 1W resistor
V2	EL519/6KG6A tube
	EL519 tube socket (Magnoval)

POWER SUPPLY

REF	ITEM
C1, C2, C3, C4	100µF, 350V capacitor
D1, D2	1N5404 rectifier diode
MOV1	Metal-oxide varistor, 120V
S1	Power switch 15A
S2	Power switch 6A
SLO	2A slow blow fuse
T1	Magnetek #N66A, 230VCT, 250VA
T2	Radio Shack #273-1352, 12.6VCT, 15VA
Fan	Fan for heat removal

OTHER

Tea Ball (Faraday Shield), 3″ diameter
Case (Hammond)
Miscellaneous hardware

not currently being used. The power supply is tucked away in the back half of the Hammond case (*Photo 9*), and the controls are mounted in the rear (*Photo 10*).

Even though the oscillator circuit is astoundingly simple, I suggest building it first on a nice big piece of wood so you can make sure it's working before squeezing it into some small space and integrating it with the audio stages. This will save you a lot of in-circuit troubleshooting time. Beware of unexpected feedback or interference, since you are operating at high frequencies where anything can happen. Make sure you decouple the power supplies to prevent the strong oscillations from causing more interference and annoying 60Hz hum.

CONCLUSIONS

This project was a very fun and interesting one for me. Not only does it demonstrate the feasibility of massless audio reproduction and omnidirectionality, but also does it in an elegant, almost sci-fi way. I highly encourage the real audiophiles out there to try to bring this idea up to the highest sound quality standards most audiophiles hold. I didn't have the money for expensive audio capacitors, nor did I have the expertise necessary to tweak this project to perfection, but I hope someone out there will!

Ultimately, the ion tweeter would operate not just at tweeter frequencies, but also at midrange and maybe even upper woofer frequencies. I would love to see a two-way stereo using a subwoofer and plasma speaker. In fact, I'm soon planning to build an ion speaker using the powerful 813 transmitter tube to try to accomplish this goal.

My senior project involved using a field effect transistor as an on/off switch in place of the lossy vacuum tube to increase the power efficiency of the oscillator section. Increasing the power efficiency was the first step in increasing the size of the flame, since it's better not to throw away massive amounts of heat. In theory, it shouldn't matter what kind of device is being used to generate the corona, because the audio character is set by the preamp stages.

RECOMMENDATIONS

- Redesign preamp stage for better linearity, dynamic range, and higher gain for compatibility with RCA-level signals.
- Use power supply with less DC ripple.
- Build a ramp-up circuit for the power supply to prevent large inrush current when turning on the supply.
- Try a better RF choke for less power line interference.
- Try to build an auto-start for the corona by injecting an impulse of current through the tube.
- Try multiple tubes in parallel for a larger flame and wider frequency response. ❖

Capacitor ESR Tester
The good, the bad, and the leaky . . .

How about an in-circuit capacitor tester to take the strain out of tracking down faulty capacitors? No need to unsolder any capacitor, simply check it in-circuit, from thousands of microfarads down to a hundred nanofarads. In most cases, parallel coils or low value resistors are no problem. Even shorted caps may be revealed in-circuit and polarity is irrelevant for the tester. High ESR? Replace! **Design by Flemming Jensen**

The most significant property of a capacitor is its capacitance, but besides that there's another important factor, namely the so-called ESR, or Equivalent Series Resistance. An ideal capacitor is a purely reactive component with a 90° phase angle between current and voltage. In the real world, however, a capacitor needs to be modeled as an ideal capacitor in series with a resistor representing the losses introduced by the component. The equivalent circuit is shown in *Fig. 1.* Sure, you can measure capacitance with a capacitance meter, which is pretty common nowadays, but unfortunately this test won't tell you anything about the capacitor's quality—you need to know the ESR as well.

Over time, electrolytics tend to dry out, which will raise their ESR and inevitably the voltage drop inside the capacitor. Evidently, the pure reactance X_c cannot produce heat, due to the 90° phase shift between voltage and current, but the ESR can, and in switching circuits the resultant heat will cause a further degradation of the capacitor's quality, i.e., a further rise in ESR. It's fairly common to find electrolytics that on the face of it have lost only just a few percent of their rated capacity although their ESR is in the hundred ohms range. Obviously, such a component acts as a load just running hot and wasting a lot of energy.

THE MEASURING PRINCIPLE

The capacitor under test, C.U.T., is fed with a 100kHz constant-current squarewave signal. The ESR value is determined by measuring the AC voltage drop across the C.U.T. If the capacitance is high enough compared to the frequency, the voltage drop over the internal reactance is negligible and the drop is caused only by the ESR. This voltage is converted to DC and fed to the voltmeter section.

AC to DC conversion of a 100kHz signal in the millivolts range presents a real design challenge. Furthermore, the conversion needs to be as linear as possible because you want to use an ordinary 200mV DVM readout. It goes without saying that an ordinary diode rectifier will not suffice, and an active diode rectifier with opamps will have a hard time working at 100kHz and a few millivolts.

The solution we came up with is a synchronous rectifier—essentially a polarity changer controlled by the same generator that supplies the 100kHz test signal. This circuit works surprisingly well and is cheap, too!

A simplified version of the circuit is shown in *Fig. 2.* Here, the C.U.T. is assumed to be a 100µF with an ESR of 10Ω. As shown, the reactance is negligible and the ESR, which is purely resistive, is dominating. Although this principle works well, further reduction of the reactive influence is called for.

Figure 3 shows an example in which the C.U.T. is 0.1µF cap whose ESR is 0Ω. As mentioned, we use a relatively high frequency to make the reactance negligible while enabling even the smallest electrolytics such as 0.1µF to be tested. For this it is necessary to re-

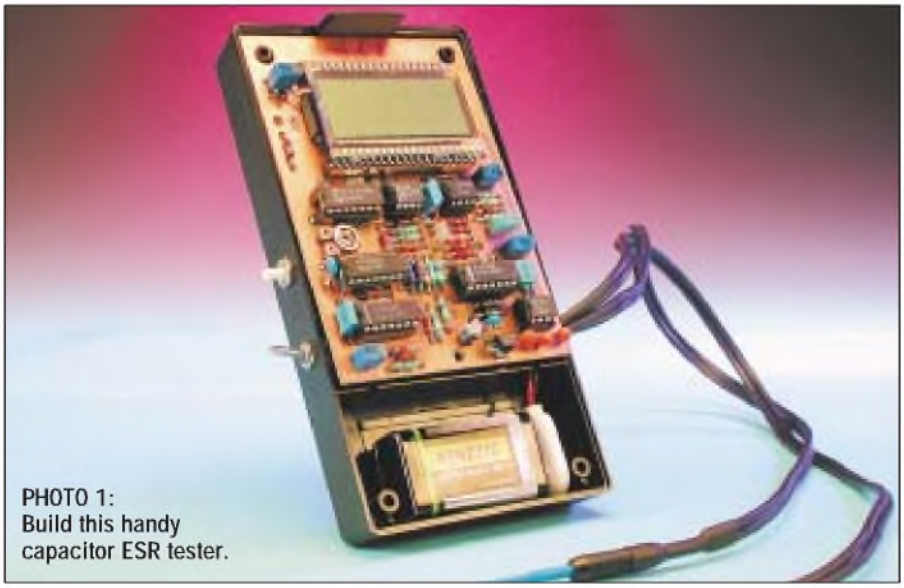


PHOTO 1: Build this handy capacitor ESR tester.

duce the influence of the beginning integration of the waveform even further. The ESR is zero and the reactance is 15Ω. As you can see, the integrated waveforms presented to the differential amplifier inputs result in a sawtooth centered around 0V at the output.

After integration, in the RC network that follows, a DC level of 0V is fed to the voltmeter circuit. If the C.U.T. also represents an ESR of, say, 10Ω, the sawtooth at the output will still have the same waveform. However, it will be DC-shifted in the positive direction by an amount representing the ESR value. After integrating the sawtooth away, the output will give the proper reading of 10Ω, excluding the 15Ω reactance.

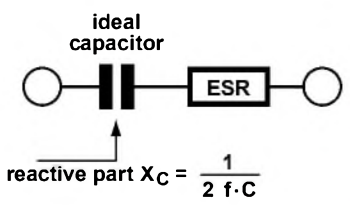


FIGURE 1: The most important property of a capacitor is its capacitance, but beside that there's another important factor, namely the so-called ESR, or Equivalent Series Resistance. A-2197-1

LOW-ESR CAP OR SHORTED CAP?

You may question whether you're testing a low-ESR cap or simply a shorted one. A simple DC ohms test is usually enough to decide this. No need to get out the multimeter—with a push of a button the ESR tester becomes a DC ohmmeter and your display should change to a higher ohm reading. If it doesn't, the odds are you have a shorted cap on your hands.

SOME PRACTICAL ESR VALUES, PLEASE?

So how high will the ESR be then? Well, that depends on where the capacitor is used, the type, the make, the voltage rating, and so forth. A 2,200µF reservoir capacitor with an ESR of 10Ω may be fine in a linear power supply, while a 2,200µF one having 1Ω ESR may be grossly inadequate in a switch-mode PSU.

In general, if a large capacitor, as in this example, reads more than 1Ω, you should be suspicious and run a comparison on a similar component. But don't worry! It won't take long before you are able to distinguish bad caps from good ones. If you regularly are repairing SMPSUs, TV sets, monitors, and so on, you will soon appreciate the ESR tester.

CIRCUIT DIAGRAM

Let's look at the circuit diagram of the Capacitor ESR Tester (*Fig. 4*). A 200kHz square-wave generator is built around IC1. This signal is divided in IC2.A, which in fact constitutes our bipolar 100kHz test signal generator. Series resistors R6 and R3-P3 on the Q and $\bar{Q}$ outputs of IC2.A give the generator a

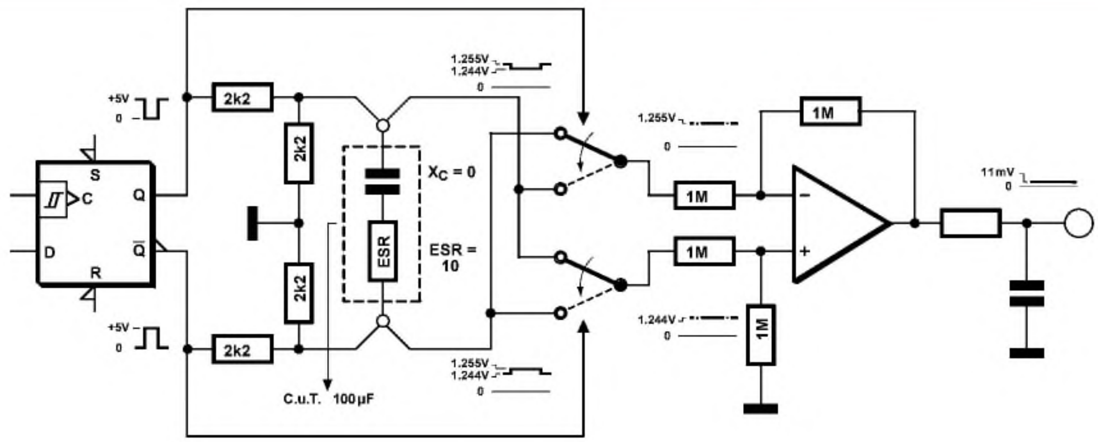


FIGURE 2: Illustrating the principle of operation. Assuming that the C.U.T. is a 100µF capacitor with an ESR of 10Ω, the reactance is negligible and the ESR, which is pure ohmic, is dominating. A-2197-2

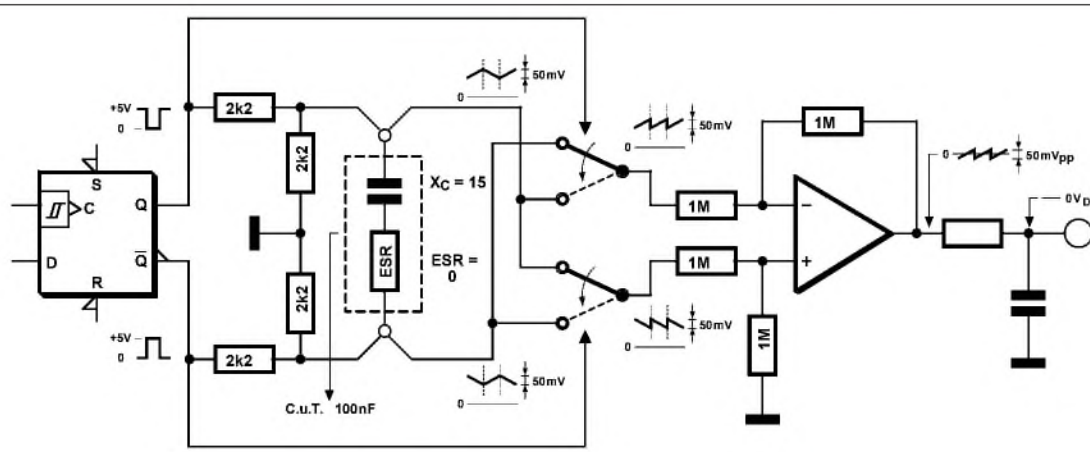


FIGURE 3: Second hypothetical test: C.U.T. is a 0.1µF capacitor with an ESR of 0Ω. A-2197-3

high output resistance compared to the low ESR, and, essentially, make the generator act as a 100kHz, balanced, constant-current generator.

The voltage drop across the C.U.T. is taken to IC3, four bilateral switches coupled as a controlled polarity changer, changing polarity in sympathy with the outputs of IC2.A. This enables IC3 to act as a (rudimentary) ADC. IC4.A, a differential amplifier, converts the differential signal into a single-ended signal, i.e., one which is referenced to ground.

IC4.B amplifies the signal such that it can be applied to a 200mV voltmeter. IC9 is the voltmeter IC. Here the ICL7106 is used with an LCD, all in a standard configuration. The LM358 in position IC8 is a comparator that tells you when it's time to change the battery. IC7, finally, generates the negative supply rail for the circuit.

As shown in the circuit diagram, the test probes carry two screened wires each. Each probe carries a signal wire (e.g., "A") and a measuring wire (e.g., "B"). More about the probes in the next section.

CONSTRUCTION

Elektor Labs designed a compact printed circuit board for the Capacitor ESR Tester. The resulting double-sided through-plated board design is shown in *Fig. 5*. As appropriate for a test instrument, the board is designed such that all adjustment points are easily accessible, in this case, from the sides of the board (multiturn presets P1, P2, P4, P5) and from the top (preset P3).

Although the construction of the board follows standard practice (of which the main maxim is: work carefully), a few things should not be left unmentioned. First, the circuit board has a screening ground plane at the component side, so you should take care to avoid short-circuits by solder blobs or solder hairs between component terminals and the ground plane. Second, ascertain, check, and double-check the polarity of any polarized component, in particular the tantalum capacitors in positions C3, C4, and C5. Tantalum capacitors when reverse polarized have a nasty habit of exploding and emitting hideous fumes.

Finally, we recommend using sockets

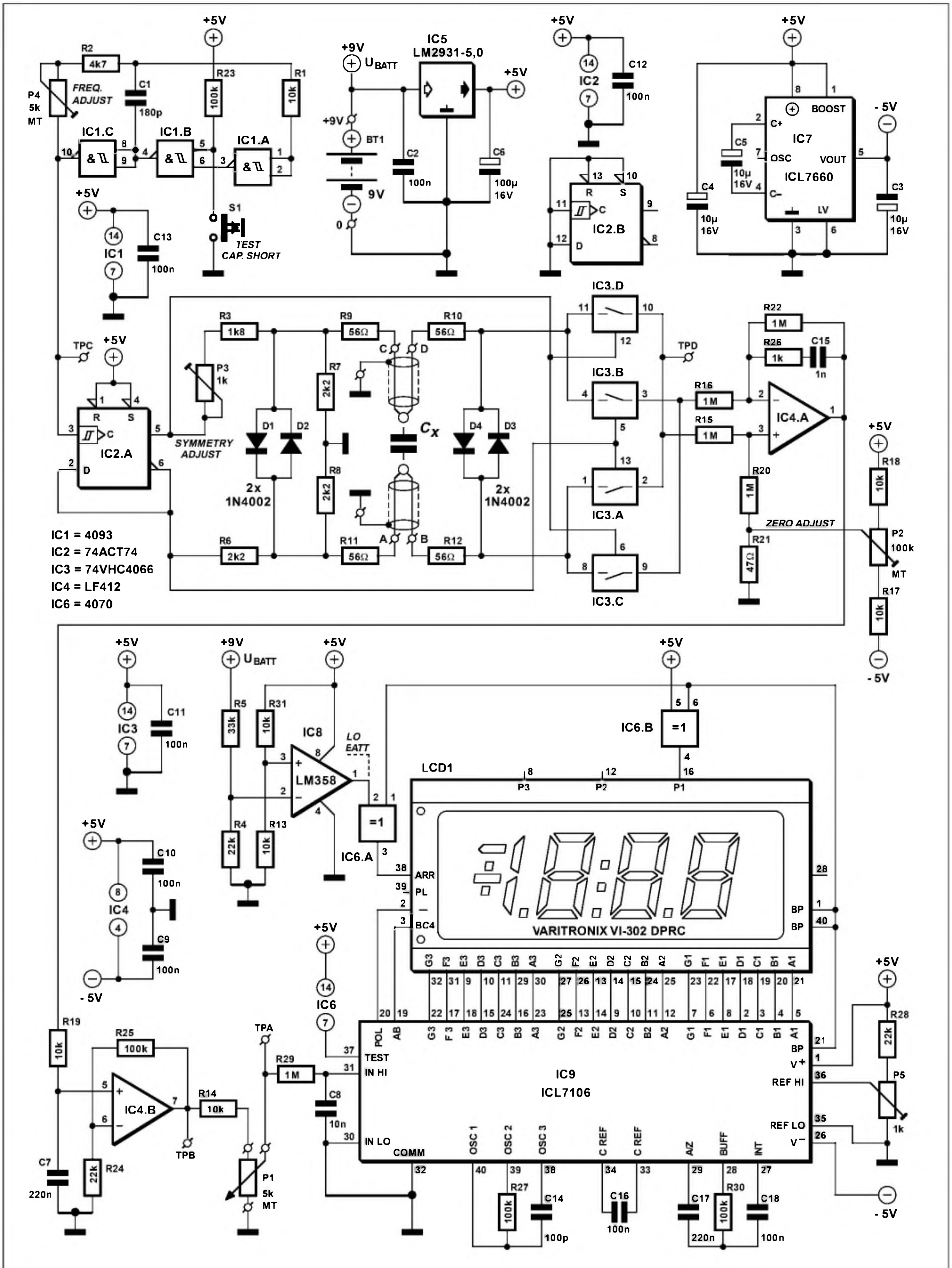


FIGURE 4: Circuit diagram of the Capacitor ESR Tester. C_X is the capacitor under test.

A-2197-4

for all ICs (except IC9) and the LCD. The latter is easily made by cutting a 40-pin IC socket in two (lengthwise) and using the two 20-way socket strips. You should drill small holes in the two long sides of the ABS case to allow P1, P2, P4, and P5 to be adjusted from the outside.

Regarding the probes, their basic construction is illustrated in *Fig. 6*. These two wires are soldered together as close to the probe tip as possible. In this way the voltage drop along the signal wire will not add to the measurement. The screening ensures that the test leads do not pick up noise, and that you maintain a stable zero adjustment.

COMPONENTS LIST

Resistors:
R1, R13, R14, R17–R19, R31 = 10kΩ
R2 = 4kΩ7
R3 = 1kΩ8
R4, R24, R28 = 22kΩ
R5 = 33kΩ
R6–R8 = 2kΩ2
R9–R12 = 56Ω
R15, R16, R20, R22, R29 = 1MΩ
R21 = 47Ω
R23, R25, R27, R30 = 100kΩ
R26 = 1kΩ
P1, P4 = 5kΩ multiturn preset, vertical mounting, side adjust (Bourns 3266X, Farnell #347-747)
P2 = 100kΩ multiturn preset, vertical mounting, side adjust (Bourns 3266X, Farnell #347-784)
P5 = 1kΩ multiturn preset, vertical mounting, side adjust (Bourns 3266X, Farnell #347-723)
P3 = 1kΩ preset, horizontal mounting

Capacitors:
C1 = 180pF
C2, C9–C13, C16, C18 = 100nF
C3–C5 = 10µF 10V radial
C6 = 100µF 16V radial
C7, C17 = 220nF
C8 = 10nF
C14 = 100pF
C15 = 1nF

Semiconductors:
D1–D4 = 1N4002
IC1 = 4093
IC2 = 74ACT74 PC
IC3 = 74VHC4066
IC4 = LF412-CN
IC5 = LM2931-5,0
IC6 = 4070
IC7 = ICL7660
IC8 = LM358-N
IC9 = ICL7106-CP

Miscellaneous:
LCD1 = 3.5 Digit LCD with LO-BATT indicator, e.g., Varitronix VI-302 DPRC (Farnell #478-660)
S1 = pushbutton, 1 make contact
Batter holder
On/off switch
Two miniature probes, e.g., Hirschmann PRUFI (Farnell #523-483)
Length of two-core screened cable
ABS enclosure with LCD window and battery compartment, e.g., Multicomp type BC4, (Farnell #645-758)
40-pin IC socket cut in half (see text)

THE ESR TESTER AS AN ADD-ON

The most costly parts in the circuit are the display and the 7106 A-D converter. You can save money by deciding to use the ESR Tester as an add-on for an existing digital multimeter (DMM). Switch the multimeter's range selector to the 200.0mV/DC position and connect the inputs to GND and the wiper of P1. You should not be tempted to supply the ESR Tester from the multimeter's battery.

Remember, the ESR Tester has its output referenced to ground, so if you run it off the multimeter's battery the Tester will have its battery minus connected to the input common terminal, which is far from advisable. Use a separate battery for the ESR Tester to avoid any problems. Or if you really want to use just one battery, give the ESR Tester an add-on 9V battery, connecting the ESR Tester's regulated +5V to the plus terminal of the multimeter's battery connector and the ESR Tester's –5V to the multimeter's minus terminal.

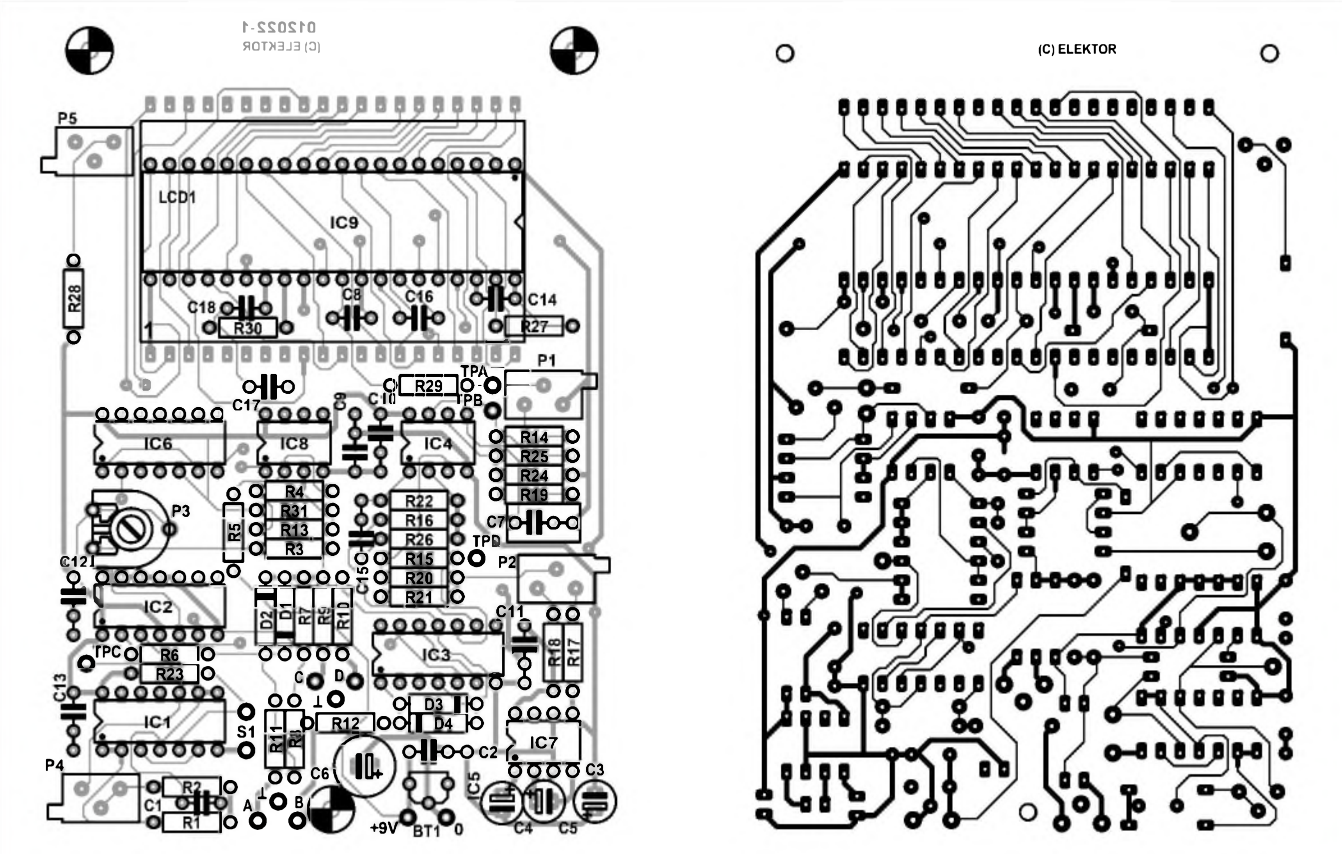
A FEW WORDS OF WARNING

Though the ESR Tester has diode-protected inputs, it is still a good idea to discharge any largish capacitors you want to test. Some reservoir capacitors in power circuits contain so much energy that the protection circuit may burn out. If this should happen, the defective components are usually to be found in

HOW DOES ESR INFLUENCE CIRCUIT BEHAVIOR?

In (fast) switching circuits, a low ESR may be crucial for proper circuit behavior. For example, in a TV set, high capacitor-ESR may lead to inability to quit stand-by mode, incorrect picture height or width, synchronization problems, interference or hum bars. In Switch Mode Power (SMPSUs) supplies, high ESR caps may lead to blown semiconductors, blown fuses, or no startup. In power circuits, a rising ESR will make the capacitor warm up, leading to even higher ESR and eventually circuit breakdown.

The usual method to troubleshoot these problems involves soldering out the capacitors, measuring the capacitance, and soldering the good ones back in. A tedious task, but what's even worse, ailing capacitors often don't show a low capacitance, are soldered back in again, and then the troubleshooting becomes really time consuming.



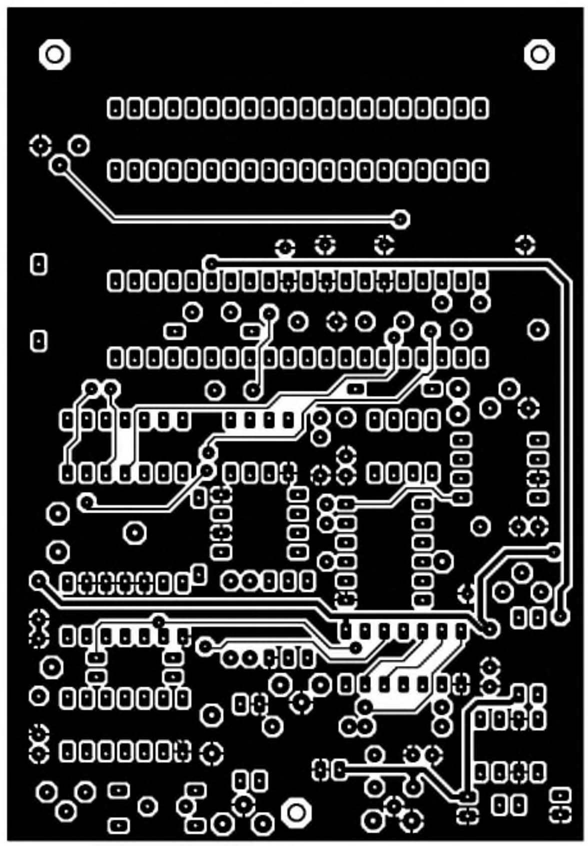


FIGURE 5: Copper track layout and component mounting plan of the PCB designed for the instrument. Double-sided, through-plated board, available ready-made.

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the protection circuit alone. The remedy should therefore be pretty straightforward and inexpensive.

ESR TESTER ADJUSTMENT

Before adjusting the instrument, be sure that you have a regulated +5V from IC5 and −5V from IC7. If you don't, you'll need to troubleshoot your circuit board.

1. Start with the voltmeter circuit. You should disconnect P1 at this point. Connect a known, accurate voltage source of less than 200mV to point TPA (test point A) and adjust P5 until the LCD shows the right value. Remove the voltage source. Connect TPA to TPB, short the test leads together, and adjust P2 for a "000.0" reading. Remove the connection. Reconnect P1.
2. Connect a frequency counter or an oscilloscope between TPC and GND. Adjust P4 for 200kHz counter reading or 5μs period time on the oscilloscope.

Connect the test leads to a 10Ω resistor. Connect an oscilloscope (in AC mode) between point TPD and GND. Turn P3 (symmetry adjust) so that the two half cycles line up and produce a straight line. Adjust P1 for a "10.0" DVM reading.

If you do not have a counter or a

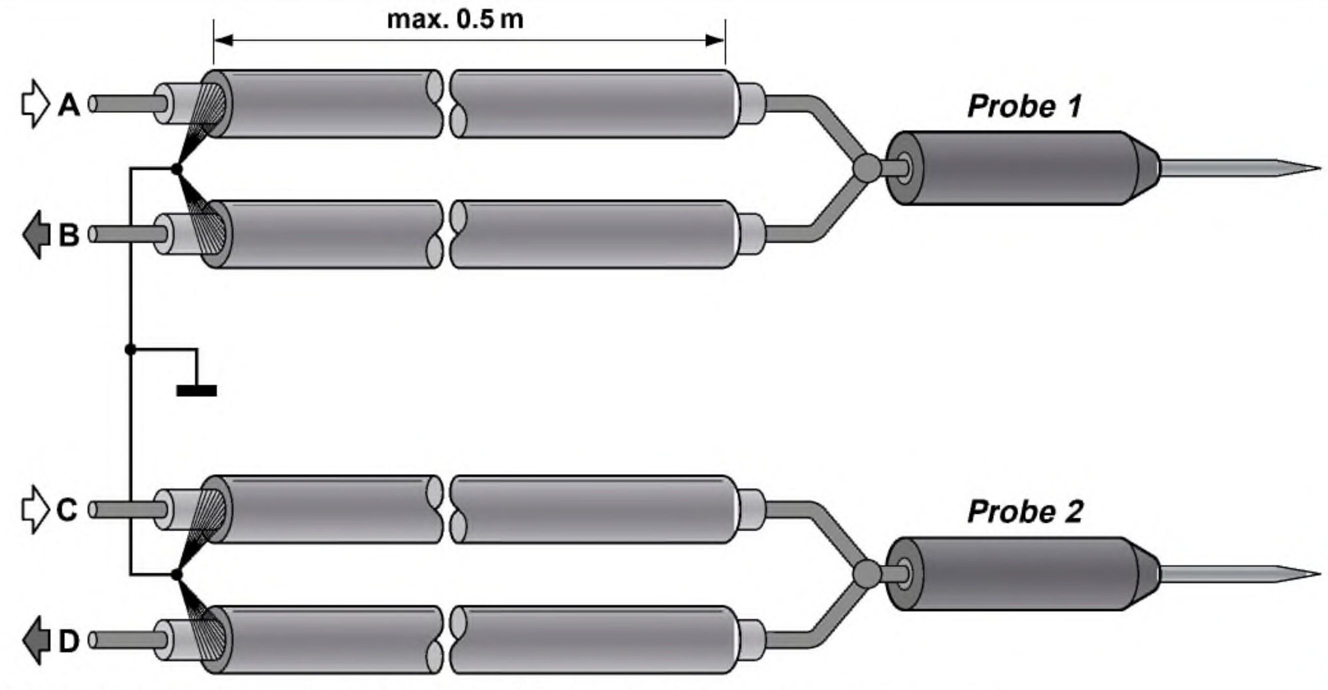

FIGURE 6: Here's how to make the four-wire test leads between the probes and the instrument proper.

scope available, turn P3 and P1 to the center of their travel. To ensure that the ESR Tester works properly, you can connect different (known, good) capacitors in series with different resistors and have these simulate capacitor ESR.

COMPONENT CONSIDERATIONS

The LF412 in position IC4 is a good choice for the differential amplifier. Since we are dealing with high-frequency signals in the millivolts range, low drift, low offset, and high bandwidth are crucial. Many different opamps have been tested but most resulted in DC drift problems. The LF412 emerged as a good, low-cost choice causing minimal drift.

IC5, then, is a 5V regulator that works just fine at a voltage drop less than 600mV and so ensures long battery life. This regulator enables the circuit to keep working down to a battery voltage of less than 6V. IC2, a 74ACT74, is capable of delivering enough current at 100kHz to produce a nice clean square wave. IC3 is a high-speed (VHC) version of the well known 4066. Compared to the common 4066, the effect of unwanted reactance is halved. For best performance, you should use the specified components, but all in all, quite acceptable performance is still achieved if you use an ordinary 4066 for IC3, and a 74HCT74 for IC2. ❖

RELATED WEBSITES:
www.awiz.com/cwinfo.htm
www.flippers.com/esrkttxt.html

Printed circuit cards for this project 012022-1 are available from Old Colony (PO Box 876, Peterborough, NH 03458, 603-924-9464, Fax: 603-924-9467, custserv@audioXpress.com) for $27.50 plus postage.

A Hybrid Stereo $90 Control

For a modest cost, this control unit offers lifelike, concert-hall performance, featuring cascaded FETs in the phono section and high-perveance tubes in the line amplifier section. **By Joseph Norwood Still**

This hybrid-design control unit (*Photo 1*) has a plug-in-the-wall 24V AC power transformer ($5.50), a chassis-mounted transformer ($5.99), a vacuum tube line amplifier, and an optional FET phono amplifier. The wall-wart power transformer is remotely located from the control unit chassis to ensure low noise operation. This transformer provides power to operate a 25V AC (input) to 120V AC (output) chassis-mounted power transformer and a full-wave, bridge-rectifier to furnish power to the series-connected filaments of the line amplifier. The 120V AC output "feeds" a full-wave, voltage-doubler, bridge-rectifier and provides 220V DC output.

Because the wall-wart transformer delivers most of the power to operate the control unit, the undesired magnetic noise of this transformer is isolated from the control unit chassis. The chassis-mounted power transformer operates at a low power level and thus generates a low level of magnetic noise.

The control unit (*Photo 2*) has a two-section, six-position selector switch. Five positions are used for line-level operation, while you can use the sixth position for a phono amplifier. Separate right- and left-channel volume controls are used, instead of a balance control, to prevent crossover distortion. The DPST power switch turns the control unit and remote power amplifier on or off.

ABOUT THE AUTHOR
Joseph Norwood Still, retired from the electronics industry, is designing affordable high-quality audio amplifiers for the dedicated audiophile. This is a hobby he thoroughly enjoys and is especially rewarded with many pleasant interchanges with dedicated, resourceful audiophiles. He resides in Bel Air, Md.

The control unit operating in the line-level or phono mode with volume control "full-on" is "dead-quiet," and you cannot detect noise with your ears inches away from the speakers. The realism of sound reproduction of the control unit with the phono and line-level amplifiers operating is excellent.

Note: The control unit features two entirely different designs (*Table 1*). The first features a traditional design using paralleled 5687s with a voltage gain of eight. The second design features the 12B4A with a voltage gain of 2.4. When operating in the CD, phono, or tuner modes, the 12B4A's low voltage gain is able to drive a power amplifier to its full output. The output signal of my phono/line amplifier is greater than the output signal of a Sony CD, Model CA9ES when "fed" through the line amplifier.

The use of the 12B4A and its low voltage gain provides a more realistic transparent sound, lower distortion, and much higher headroom from signal overload than the more traditional 5687. This gives substantial "headroom" at the input and output of the tube for reproduction of music peaks with no danger of clipping. The low amplification factor, low plate resistance, and high cathode bias (16V DC) make the 12B4A tube a great line amplifier and explains its excellent performance.

I have provided separate descriptions for the 5687 and 12B4A line amplifier designs, so you can opt to build the line amplifier that best matches your own preference and audio system.

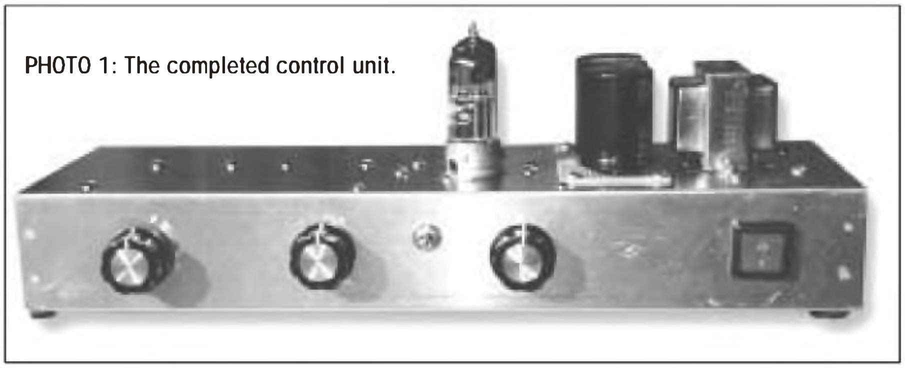
PHOTO 1: The completed control unit.

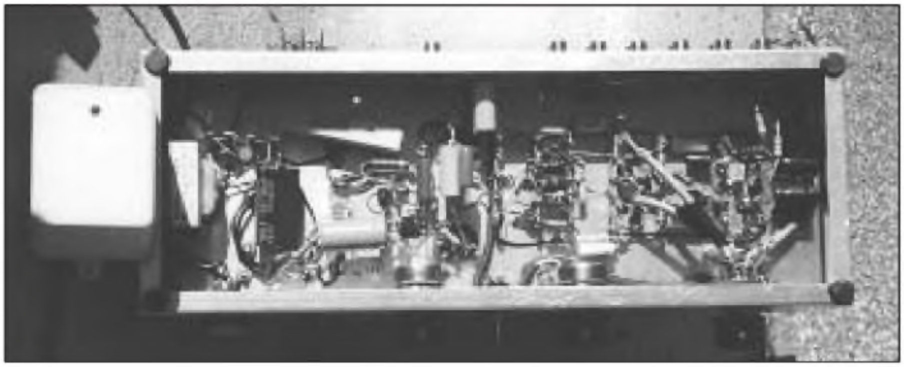
PHOTO 2: Inside view.

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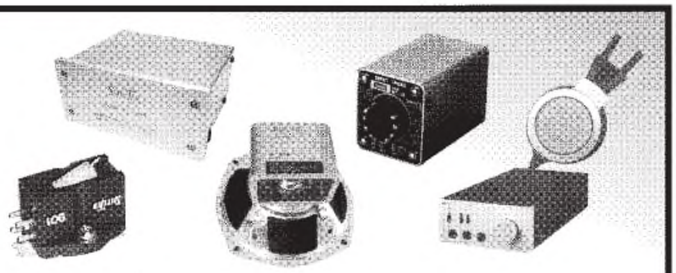

■Cartridge

Model	Price (US$)
Denon DL-102 (MONO)	150
Denon DL-103 (STEREO)	200
Denon DL-103R (STEREO)	250
Denon DL-103 PRO (STEREO)	350
Shelter Model 501 II (CROWN JEWEL REFERENCE)	750
Shelter Model 901 (CROWN JEWEL SE)	1,400

Postage (Air Economy)	
Area I $18	China, Korea, Hong Kong, Taiwan
Area II $22	Singapore, Malaysia, Indonesia
Area III $27	North America, Oceania, Europe
Area IV $34	Africa, South America

These Area I ~ IV are for all products except book.

■Japanease Audio Book

Postage $15

Title		Price (US$)
Attractive Tube Amps Vol. 1&2	(Isamu Asano)	30 each
The Joy of Vintage Tube Amps 1&2	(Tadaatsu Atarashi)	30 each NEW
Direct & Indirect Tube Amps	(Kiyokazu Matsunami)	40 NEW
SE Amps by Transmitting Tubes	(Kouichi Shishido)	50
The Remembrance of Sound Post	(Susumu Sakuma)	30
Classic Valve	(Hisashi Ohtsuka)	40
MJ Selected 300B Amps	(MJ)	30
Top-Sounding Vintage Power Tubes	(Stereo Sound)	30
Output Trans of The World	(Stereo Sound)	30
20TH CENTURY OF AUDIO	(Stereo Sound)	30
Vintage Speaker Units	(Stereo Sound)	30 NEW
Tube Amp Craft Guide	(MJ)	30

■MC STEP UP TRANS

Model	Specifications			Price (US$)	Postage**
	Pri.Imp(Ω)	Sec.Imp(kΩ)	Response		
Shelter Model 411	3～15	47	20Hz～50kHz	980	Area I $25
Jensen JE-34K-DX	3	47	20Hz～20kHz	550	Area II $30
Peerless 4722	38	50	20Hz～20kHz	300	Area III $40 Area IV $50

**Air Economy

■STAX

Model	Price (US$)
OMEGA II System (SR-007+SRM-007t)	Ask
SRS-5050 System W MK II	Ask
SRS-4040 Signature System II	Ask
SRS-3030 Classic System II	Ask
SRS-2020 Basic System II	Ask
SR-001 MK2 (S-001 MK II +SRM-001)	Ask

■Speaker

Model	Specifications					Price* (US$)	Postage** (US$)			
	D (cm)	Ω	Response	db	w		I	II	III	IV
Fostex FE208Σ	20	8	45Hz～20kHz	96.5	100	296	62	74	120	156
Fostex FE168Σ	16	8	60Hz～20kHz	94	80	236	42	50	73	98

*Price is for a pair **Air Economy

■TANGO TRANS (ISO) 〔40models are available now〕

Model	Specifications				Price (US$)	Postage** (US$)			
	W	Pri.Imp(kΩ)	Freq Response	Application		I	II	III	IV
XE-20S (SE OPT)	20	2.5, 3.5, 5	20Hz～90kHz	300B,50,2A3	396	47	56	84	113
U-808 (SE OPT)	25	2, 2.5, 3.5, 5	20Hz～65kHz	6L6,50,2A3	242	42	50	73	98
XE-60-5 (PP OPT)	60	5	4Hz～80kHz	300B,KT-88,EL34	620	62	74	115	156
FX-40-5 (PP OPT)	40	5	4Hz～80kHz	2A3,EL34,6L6	320	47	56	84	113
FC-30-3.5S (SE OPT) 〔XE-60-3.5S〕	30	3.5	20Hz～100kHz	300B,50,PX-25	620	62	74	115	156
FC-30-10S (SE OPT) 〔XE-60-10SNF〕	30	10	30Hz～50kHz	211,845	620	62	74	115	156
X-10SF 〔X-10S〕	40	10W/SG Tap	20Hz～55kHz	211,845	1160	90	110	180	251
NC-14 (Interstage)	—	〔1+1：1+1〕5	25Hz～40kHz	〔30mA〕 6V6(T)	264	30	40	50	70
NC-16 (Interstage)	—	〔1+1：2+2〕7	25Hz～20kHz	〔15mA〕 6SN7	264	30	40	50	70
NC-20F 〔NC-20〕 (Interstage)	—	〔1：1〕5	18Hz～80kHz	〔30mA〕 6V6(T)	640	42	50	73	98
NP-126 (Pre Out)	—	20,10	20Hz～30kHz	〔10mA〕 6SN7	264	30	40	50	70

Price is for a Pair

**Air Economy

■TAMURA TRANS (All models are available)

Model	W	Pri.Imp(kΩ)	Freq Response	Application	Price (US$)	I	II	III	IV
F-7002 (Permalloy)	10	3.5	15Hz～50kHz	300B,50	836	60	70	110	145
F-7003 (Permalloy)	10	5	15Hz～50kHz	300B,50	836	60	70	110	145
F-2013	40	10	20Hz～50kHz	211,242	786	70	84	133	181
F-5002 (Amorphous)	8	3	10Hz～100kHz	300B,2A3	1276	65	80	120	160

Price is for a Pair

**Air Economy

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LINE AMPLIFIER (USING THE 5687)

The line amplifier (*Fig. 1*) uses two 5687 tubes whose heaters are connected in a series string configuration. The heaters are operated at 20V DC, supplied by a well-filtered, full-wave bridge-rectifier. The plates of the 5687 are operated from a supply voltage of +220V DC/ 10mA. The heater of the 5687 is operated at a reduced voltage of 10V DC to obtain low-noise operation from this stage. The normal heater power for the 5687 is excessive, operating at the same power level as a 6L6.

The circuit of the line amplifier is simple and straightforward. The 5687 is especially suited for use as a line amplifier with its 6V DC cathode bias and its 1.1% distortion at 20V RMS output. This provides considerable headroom at both the input and the output of the tube and ensures no clipping will occur during reproduction of music peaks.

The noise of the line amplifier is 0.3mV with volume control (R1) full on and the input open, non-shorted. (I measured the noise with a digital multimeter (DMM), Radio Shack, 22-168A.) All noise and ripple measurements throughout this article use a DMM with shielded test leads. The distortion from 40Hz to 20kHz of the line-level stage is typically 0.28% at 5V RMS output, while at 3V RMS it is 0.18% (see *Table 2* for more distortion data).

The frequency response of the line amplifier is flat from 10Hz to 20kHz; feeding a 6′ audio cable (Radio Shack 15-1505), the frequency response is flat from 10Hz to 18kHz. The high-frequency operation of the line amplifier is enhanced by paralleling capacitors C1 (0.47µF) and C2 (.047µF).

The resistors R2, R5, and R6 are added to ensure stability. The 22kΩ plate resistor and 1.2kΩ cathode resistor provide a low distortion output. A separate 100k volume control is used for each stereo channel to ensure maximum stereo separation and minimum crossover distortion.

LINE AMPLIFIER (USING THE 12B4A)

The line amplifier (*Fig. 2*) uses high-perveance 12B4A tubes whose heaters are connected in a series string configuration. The heaters are operated at 24V DC, supplied by a well-filtered, full-wave, bridge rectifier. The plates of the 12B4A are operated from a supply volt-

TABLE 1

5687 ELECTRICAL CHARACTERISTICS		12B4A ELECTRICAL CHARACTERISTICS	
Ebb	220V DC	Ebb	220V DC
Eb	115V DC	Eb	115V DC
Ib	5mA	Ib	5mA
Rk	1.2kΩ (+6V DC)	Rk	3.3kΩ (+16V DC)
RL	22kΩ	RL	22kΩ
A.F.	18	A.F.	6.5
V.G	8	V.G.	2.4

TABLE 2
5687 AND 12B4A LINE AMPLIFIER—TOTAL HARMONIC DISTORTION

VOLTS OUT (RMS)	20HZ 5687	20HZ 12B4A	40HZ TO 20KHZ 5687	40HZ TO 20KHZ 12B4A
1	0.1%	0.09%	0.08%	0.08%
3	0.22%	0.15%	0.18%	0.11%
5	0.34%	0.22%	0.28%	0.22%
8	0.5%	0.34%	0.44%	0.28%
10	0.6%	0.4%	0.56%	0.36%
15	0.88%	0.6%	0.82%	0.56%
20	1.2%	0.9%	1.1%	0.8%

Note: Clipping for the 5687 and 12B4A occurs above 20V RMS.

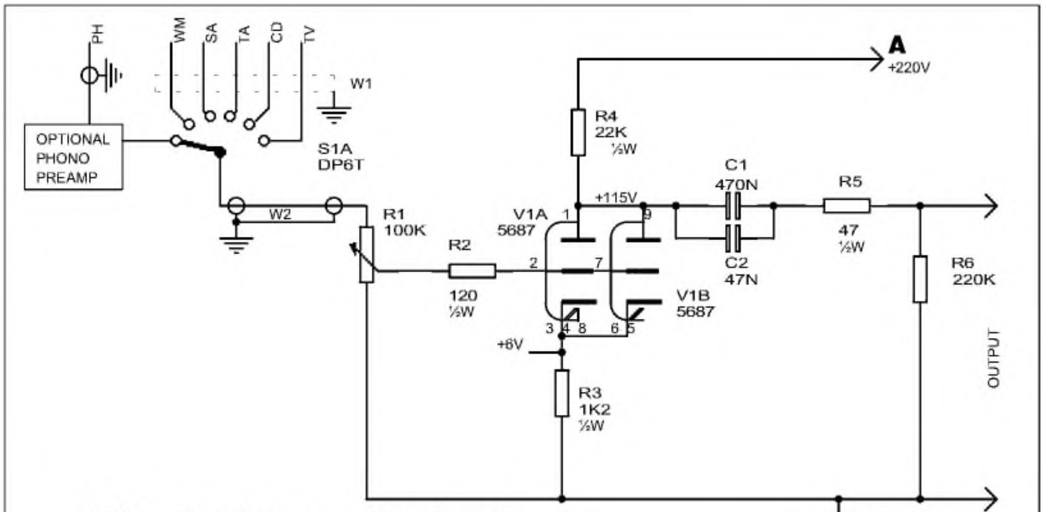


FIGURE 1: The 5687 line amplifier schematic.

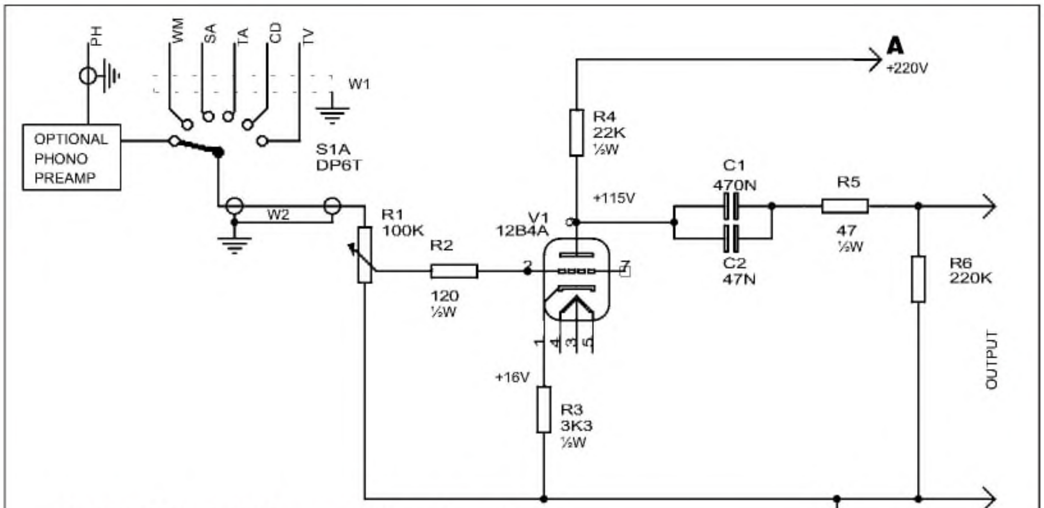


FIGURE 2: The 12B4A line amplifier schematic.

age of +220V DC/10mA.

The circuit of the line amplifier is simple and straightforward, and the excellent performance is related to the outstanding features of the 12B4A tube in this application. It has a plate curve that jumps straight up from the base line and is very linear, as its low distortion characteristics verify. You can apply the axiom, a straight wire with gain, to this tube because of its low voltage gain.

In my opinion, the sonic performance of the 12B4A as a line amplifier is superior to the 5687, 6SN7, or 6H30. The 12B4A is especially suited for use as a line amplifier with its 16V DC cathode bias and its 0.8% distortion at 20V RMS output. This provides incredible headroom at both the input and the output of the tube and ensures no clipping will occur during reproduction of music peaks. The use of this tube will provide total transparency at the line-level stage and will permit you to concentrate on the power amplifier and the input stage preceding the line amplifier for further refinement of your audio system.

The noise of the line amplifier is 0.1mV with volume control (R1) full on and the input open, non-shorted. The distortion from 40Hz to 20kHz of the line-level stage is typically 0.22% at 5V RMS output, while at 3V RMS it is less than 0.11% (see *Table 2* for more distortion data).

The frequency response of the line amplifier is flat from 10Hz to 20kHz; feeding a 6′ audio cable (Radio Shack 15-1505), the frequency response is flat from 10Hz to 18kHz. The high-frequency operation of the line amplifier is enhanced by paralleling capacitors C1 (0.47μF) and C2 (.047μF).

The resistors R2, R5, and R6 are added to ensure stability. The 22kΩ plate resistor and 3.3kΩ cathode resistor provide a low distortion output. A separate 100k volume control is used for each stereo channel to ensure maximum stereo separation and minimum crossover distortion.

Note: If you build the phono amplifier, I recommend you use the 12B4A as a line amplifier, because its low voltage gain is more compatible with the high output of the phono amplifier.

POWER SUPPLY

The power supply (*Fig. 3*) consists of a wall-wart 120V AC to 24V AC transformer, built-in overload protection. The wall-wart secondary is rated at 830mA/24V AC, which is conservatively rated to furnish power to operate the FETs and 5687/12B4As. The wall-wart 24V AC output voltage feeds a full-wave, bridge-rectifier that provides 20/24V DC output to the heaters of series-connected 5687/12B4As. It is filtered by capaci-

TABLE 3
DRAIN SUPPLY VOLTAGE 56V DC, DRAIN LOAD RESISTOR 39K, DRAIN VOLTAGE 12V DC

	INPUT SIGNAL/1KHZ	OUTPUT SIGNAL	*SOURCE BIAS VOLTAGE	DISTORTION %
Q1-R	70mV RMS	0.3V RMS	0.99V DC	0.095
Q1-L	70mV RMS	0.3V RMS	1.38V DC	0.080
Q2-R	40mV RMS	0.3V RMS	0.5V DC	0.120
Q2-L	40mV RMS	0.3V RMS	0.6V DC	0.125

*Source bias variable resistor is adjusted for a drain voltage of 12V DC.

tor C1, and resistor R2 (20Ω) is a voltage dropper device. The large value of capacitor C1 reduces the noise voltage to .002V AC at the 20/24V DC output of resistor R2.

The 24V AC output of the wall-wart also feeds the 25V AC winding of chassis-mounted transformer T2. The 120V AC output of transformer T2 feeds a full-wave bridge (BR-2), voltage-doubler rectifier. Voltage doubling is accomplished by capacitors C2 and C3. The choke L1 diminishes the switching transients that occur during the rectification process.

Capacitor C4 provides additional filtering and a low noise of .004V AC at the +220V DC output. (I measured the noise with a 2μF Mylar capacitor connected in series with the DMM shielded test leads). The +56V DC (.001V AC noise) to operate the phono amplifier is obtained at the junction of zener diodes ZD-2 and ZD-3. Capacitor C7 filters the +56V DC output. Capacitors C5 and C6 eliminate AC spikes.

PHONO AMPLIFIER

FET's Widely Variable Electrical Characteristics

Important: When building FET amplifiers, it is essential to establish an environment conducive to minimum handling (static electricity) and heat contact of the FET. To ensure this occurs, install transistor sockets for the FETs Q1 and Q2. It is not necessary to have the socket mounted in the chassis—you may solder stiff wires to the socket wires, which you then solder to their respective components/terminal strips.

Note: For audio applications, the primary variable electrical characteristics of the FET is the bias control point. A variation in the bias control point affects the drain current and thus the voltage presented to the drain via the drain load resistor. This, of course, affects the distortion.

The obvious solution is to install a variable resistor in the source circuit and adjust the resistor to obtain the desired drain voltage (or more precisely, the drain voltage shown on the schematic). The phono amplifier presented here uses a variable resistor in each source element of the four FET stages. The variable resistors (R3 and R9) are pre-set to 1kΩ before they are connected in the circuit to avoid damaging FETs Q1 and Q2.

Of interest: The commercial manu-

TABLE 4
LINE AMPLIFIER PARTS LIST (DOUBLE ALL PARTS EXCEPT S1, W1, W2)

C1	0.47μF, 400V	AE	C-PD 47-400
C2	0.047μF, 400V	AE	C-PD047-400
R1	100k, audio taper, potentiometer	RS	271-1722
R2	120Ω, ½W, metal film	AE	R-M 120
R3 (5687 only)	1.2k, 1W, metal film	AE	R-E 1.2 K
R3 (12B4A only)	3.3k, 1W, metal film	AE	R-E 3.3 K
R4	22k, 1W, metal film	AE	R-E 22 K
R5	47, ½W, metal film	AE	R-M 47
R6	220k, ½W, metal film	AE	R-M 220K
J1–J7	RCA, phono jacks, female	ALEL	RCMJ
S1	Rotary switch, 2-pole, 6-position	RS	275-1386
V1 (5687 only)	5687 (NOS)	AE	5687
V1 (12B4A only)	12B4A (NOS)	AE	12B4A
X1	T-6 ½ tube socket, 9-pins	AE	P-ST9-213
W1	5-cond., 22 gauge, shielded	ALEL	5C-S22
W2	Single conductor, shielded	RS	42-2371

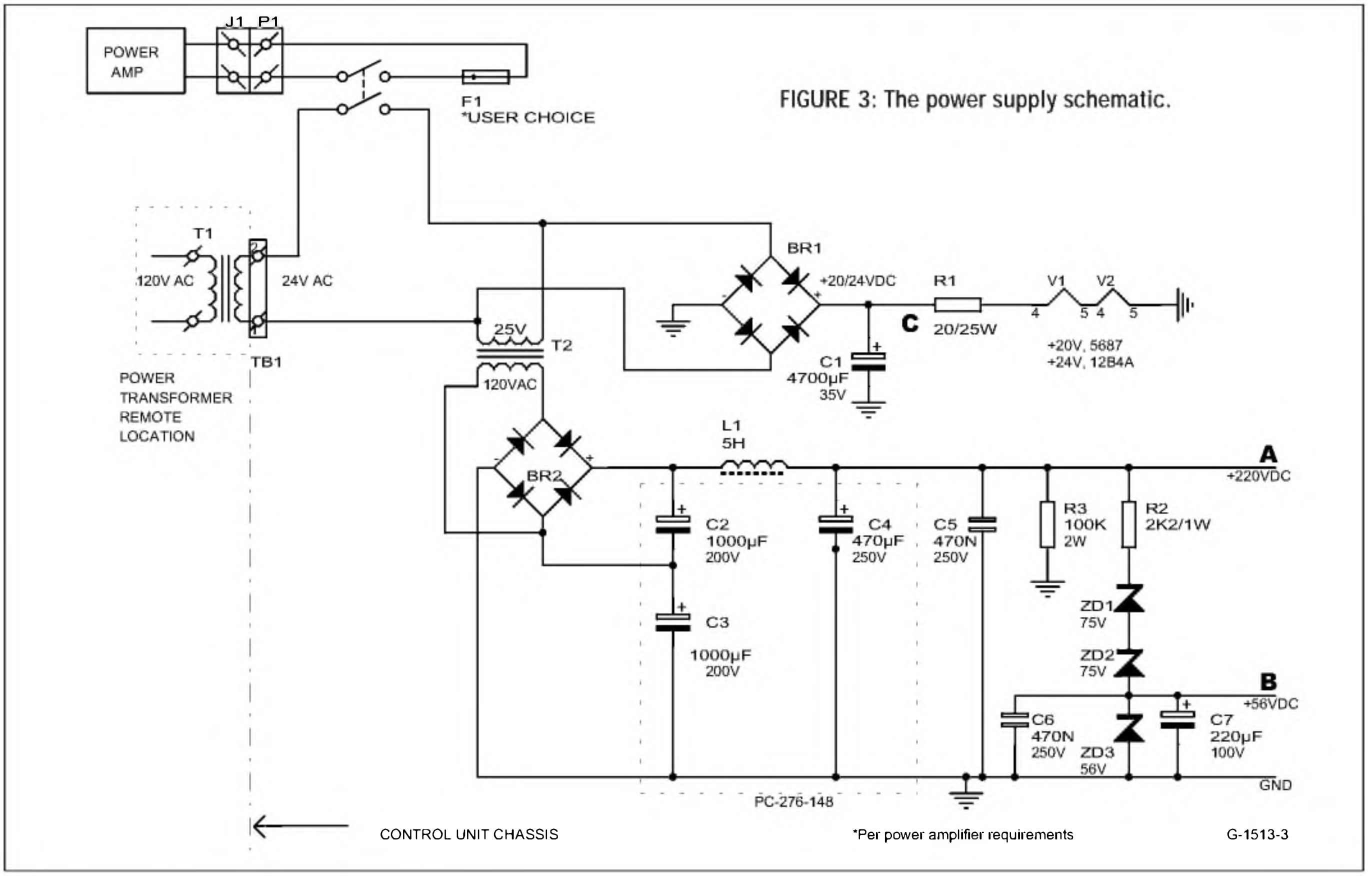


FIGURE 3: The power supply schematic.

facturer of phono amplifiers requests that hand-selected FETs have tightly controlled electrical characteristics and pay a premium price for this service. They also are required to purchase a minimum quantity of 100 FETs. When the FETs are received in-house, they are again pre-tested to ensure the desired conformity and electrical tolerance characteristics are obtained, before they are finally wired into their respective circuit.

CIRCUIT ANALYSIS

The phono amplifier (*Fig. 4*) uses readily available NTE459 FETs ($2.66). It features a non-feedback design with a passive RIAA network. The noise is less than 0.1mV, and sensitivity is .003V for 1.25V RMS output at 500Hz or 0.9V RMS at 1kHz.

The distortion of the first-stage FET is less than 0.12% from 20Hz to 20kHz with the input signal adjusted from 0.3V RMS output at all measured frequencies. The distortion of the second-stage FET is less than 0.15% from 20Hz to 20kHz with 140mV input to the gate of the second stage FET and 1.0V RMS output at the drain. These distortion measurements are made by bypassing the RIAA circuit. For distortion measurements of the 5687 and 12B4A line amplifiers, refer to *Table 2*.

The frequency response of Q1 and Q2 with inverse RIAA network connected to the input is shown from 20Hz to 20kHz (*Fig. 5*). I used an inverse RIAA network to ensure that the circuit components chosen for the RIAA network provided a reasonably flat frequency response. Although not entirely flat, the curve deviations provided a boost to the low and high frequencies. For a control unit that has no tone controls, this approach gave excellent listening results.

With .003V, 500Hz fed to the input of Q1, the output at Q1 should indicate 0.12V RMS. The loss through the RIAA network drops the 0.12V to 0.0165V at the input of Q2. The voltage at the output of Q2 is 1.25V RMS. The output of Q2 feeds the line amplifier (12B4A or 5687). The output of the line amplifier is 9V RMS (5687) and 3V RMS (12B4A) with 3mV, 500Hz at the input of the phono amplifier.

With a record playing, the output of the line amplifier is typically a swinging 1 to 10V AC (5687) and 0.6 to 4V AC (12B4A). The noise at the output of the line amplifier with the six-position, rotary switch set to the phono position is typically 0.8mV (with the 600Ω load of the audio oscillator across the phono input and the audio oscillator turned-off, simulating the phono cartridge load).

Note: The majority of commercial outboard phono amplifiers have output voltages of 0.2 to 0.3V RMS at 500Hz, and a few have outputs as high as 1.0V RMS. The phono amplifier described in this article has an output of 1.25V RMS with a 500Hz or 0.9V RMS at 1kHz with a 3mV input signal, loaded by a 100kΩ potentiometer. The signal of the phono amplifier is higher than a Sony CA9ES CD player.

The volume controls are set at the one-quarter position for living-room listening levels when in the phono mode. To obtain the same loudness level in the CD mode, you must advance the volume control to the half-on position. This confirms the high output design of this all-FET phono amplifier. For use with the typical audio system, the gain may be reduced by omitting source bypass capacitor, C8.

The individual circuit components of the phono amplifier operate as follows: C1 tailors the high-frequency output of the magnetic pickup, and R2 provides the proper load to the cartridge. Resistors R1 and R6 provide resistance decoupling. Capacitors C2 and C8 enhance the gain of the FETs Q1 and Q2. Capacitors C6, C7 and resistors R6, R7 form the RIAA network.

Capacitors C3 and C9/C10 are coupling capacitors. Capacitor C4 is required to stop unwanted high-frequency oscillation. Capacitor C5 and resistor R5 de-couple Q1 from Q2. Resistors R4 and R10 are drain load resistors, and R3 and R9 are source bias adjustable resistors.

Important: The major design problem in dealing with the widely varying electrical characteristics of FETs is the shifting bias control point. The source bias R3 and R9 are adjusted for a drain voltage that provides minimum distortion of Q1 and Q2 (*Fig. 4*). I determined that minimum distortion occurred with the drain voltage via the load resistor set at 12V DC. It will be noted the source bias of the FETs varied considerably with the drain voltage set for 12V DC.

TABLE 5
PHONO PREAMPLIFIER PARTS LIST (DOUBLE ALL PARTS)

C1	170pF, 50V, mica	ALEL	DMCP-170
C2, C8	220µF, 16V, electrolytic	RS	272-956
C3, C9	0.22µF, 100V, 5%, Mylar	ALEL	AMC-71
C4	82pF, 200V, 10%, ceramic	Mouser	539-CK05820K
C5	220µF, 100V, electrolytic	ALEL	
C6	.010µF, 100V, 2%, poly.	Mouser	140- PF2A103F
C7	.039µF, 100V, 2%, poly.	Mouser	140-PF2A393F
C10	.047µF, 100V, 10%, Mylar	ALEL	AMC-20
R1	47, ½W	ALEL	
R2	47k, ½W	ALEL	
R3, R9	2k, ½W, variable resistor, top adjust	ALEL	STP-2K
R4, R10	39k, ½W	ALEL	
R5	390, ½W	ALEL	
R6	62k, ½W	ALEL	
R7	8.2k, ½W	ALEL	
R8	1.0M, ½W	ALEL	
Q1, Q2	NTE 459	Mouser	526-NTE 459
Sockets (qty. 4)	Case TO-5, 3-pin	Mouser	573-93103

TABLE 6
POWER SUPPLY PARTS LIST

T1 (wall-wart)	120V AC to 24V AC, 830mA	ALEL	ACTX-2420
T2	120V AC to 25V AC, 450mA	RS	273-1366
L1	Choke, 5H, 5mA	ALEL	P-T155H
BR-1, BR-2	1½A, 400V AC, PIV (2 in pack)	ALEL	FWB-15
R1	20, 25W, wirewound	Mouser	280-CR25-20
R2	2.2k, 2W, metal oxide	Mouser	262-2.2K
R3	100k, 2W, metal oxide	Mouser	262-100K
C1	4700µF, 35V (radial, electrolytic)	RS	272-1022
C2, C3	1000µF, 200V (clip-in electrolytic)	ALEL	EC-1020
C4	470µF, 250V (clip-in electrolytic)	ALEL	EC-4725
C5, C6	0.47µF, 250V (radial, Mylar)	ALEL	RMC-85
C7	220µF, 100V, electrolytic	ALEL	
ZD-1, ZD-2	75V, 5W, zener diode	RS dot Com	900-2949
ZD-3	56V, 5W, zener diode	RS dot Com	900-2945
PC-1	Printed circuit board	RS	276-148
TS	Terminal strips (3-packs)	RS	274-688
J1	2 position, enclosed shell socket	RS	274-201
P1	2 position, enclosed shell socket	RS	274-202
TB-1	Barrier strip, 2 positions	RS	274-658
S1	Toggle switch, DPST	Mouser	10DS059SWD-B
Chassis box	Aluminum, 13.5 × 5 × 2	AE	P-H14444-18
Cover plate	13.5 × 5	AE	P-H14-34-18
Rubber feet	Self stick, rubber	RS	64-2342
Solder, rosin core 60/40		RS	64-006

It is obvious that calculations or SPICE is incapable of determining what the proper bias is for individual FETs. The obvious solution was a variable resistor installed in the source circuit for each of the four FETs. Adjusting the FETs to obtain 12V DC provides a high degree of assurance that these devices are operating at minimum distortion. The electrical characteristics information of the four FETs used in the phono amplifier are shown in *Table 3.* Note that the only variable of any significance is the source bias voltage.

Note: You can add a 1000µH, 14Ω, R-F choke (Mouser, 542-77F102) to the input of the phono amplifier, if needed, for R-F suppression. I did not have any problem in this area with R-F interference. The choke goes between the phono jack and 47Ω resistor.

The +56V DC/5mA power to operate the FETs is obtained at the junction of zener diodes ZD-2 and ZD-3 of the power supply (*Fig. 3*). The regulated +56V DC power "feeding" the phono amplifier has a significant impact on the sound of the control unit.

Important: I recommend that if you build FET circuits you have an audio oscillator and vacuum tube voltmeter or DMM, but you can still achieve good results with only a DMM. The FETs provide a very detailed and clean output with a solid bass response. The high-

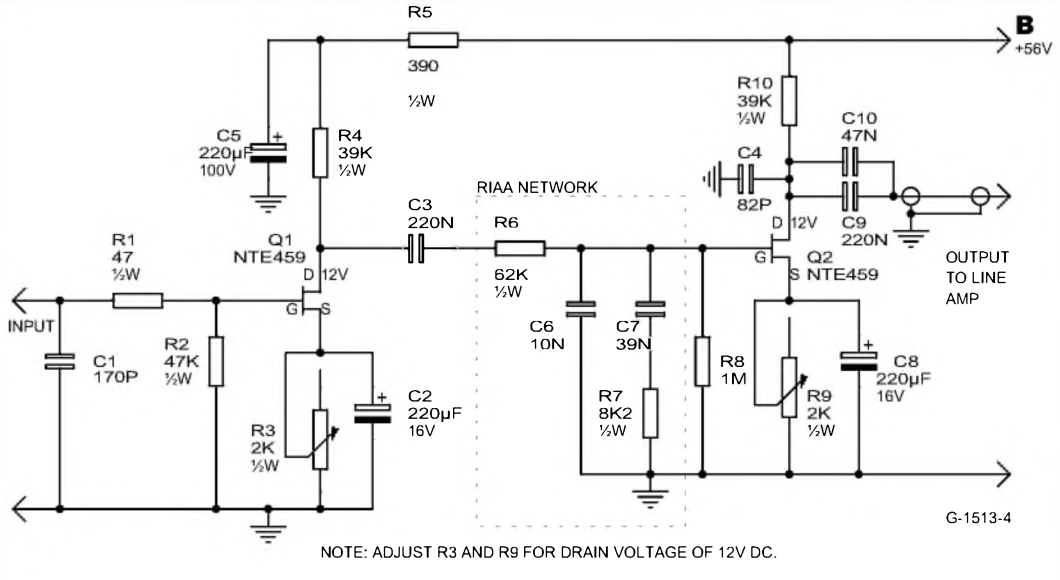


FIGURE 4: The phono amplifier schematic.

and mid-frequency bands are reproduced with exceptional clarity. FETs also have linear phase characteristics over their bandwidth for maximum signal integrity, low distortion, and high dynamics with output signals not exceeding 2V RMS. Refer to schematic diagram (*Fig. 4*) of the phono amplifier for further technical information.

The cost of the phono amplifier is $34 and includes four NTE459s. It is recommended that you order an additional four NTE459s, which cost $2.66 a piece. (**Important:** The ECG 459 is not compatible with the NTE459, so do not attempt a substitution.) The 2V RMS limited output capability of a FET is the reason FETs are not suitable for a line amplifier. However, vacuum tubes are well suited for this application, and the 5687/12B4A with its low distortion and high output voltage makes it especially appropriate. I am very impressed with the performance of the 12B4A.

CONSTRUCTION

The locations of major parts are self evident from *Photos 1* and *2*. The coupling capacitors with a black band face toward the negative side of the circuit. All the electrolytic capacitors are grounded at the negative band, except those connected in series. The high voltage electrolytic capacitors (C2, C3, C4) in the power supply section of the control unit (*Fig. 3*) are mounted on the PC board (RS-276-148).

In the phono amplifier section (*Fig. 4*), seven 5-pin terminal strips are required. Components mounted on these terminal strips are as follows (numbers indicate location order of terminal strips. Components that have one end tied to a terminal strip are not mentioned).

1. R2 (R+L), R4 (R+L), C5 (R+L)
2. R3 (R+L), C2 (R+L)
3. C6 (R), C7 (R), R7 (R)
4. C6 (L), C7 (L), R7 (L)
5. R8 (R+L)
6. R9 (R+L), C8 (R+L), C11 (R+L)
7. B+ tie points, R11 (R+L), R5 (R+L)

Note: R-Right channel, L-Left channel

It is very important that you tailor the assembly around the implementation of the eleven 5-pin terminal strips. Using the terminal strips provides a very simple step-by-step construction process. It also permits easy changeability of parts.

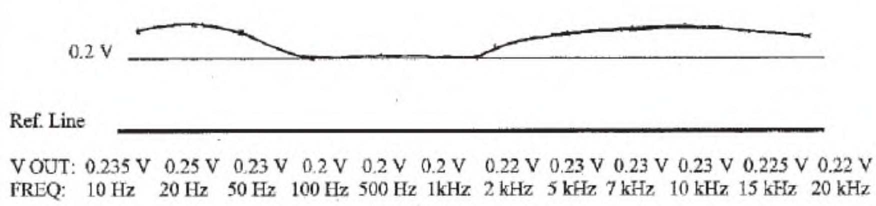


FIGURE 5: Phono preamplifier RIAA frequency response curve. Note: 0.1V RMS to input of inverse RIAA network at all frequencies and output of network connected to input of phono preamplifier. Adjust line amplifier output for 0.20V at 500Hz.

G-1513-5

Building the control unit is a fun, easy project using the 5-pin terminal strips and a "nightmare" using a PC board. Also, use point-to-point wiring and do not use a common ground. To assure simple and effective grounding of all components, use an aluminum chassis box.

It is important all audio that runs approximately 3″ or more use standard shielded microphone cable and that the shield is grounded at both ends. It will only be necessary to ground the shielded leads at the output of the phono amplifier at the end nearest the output of the amplifier. The 5-conductor shielded cable that connects to the 6-position rotary switch must be grounded at both ends. Most grounds are made to the center post, 5-pin terminal strip. I also recommend that you use chassis-mounted solder ground lugs. Add a ground post near the input of the control unit for the ground wire of the record player motor. A bottom cover plate of the chassis is required to reduce noise pick-ups.

In conclusion, this is a control unit with several unique features, the most important is its exceptional "concert hall" life-like sound. This hybrid control unit is one of the best devices I have ever built and provides many hours of listening pleasure. I hope the information on the use of FETs was helpful. ❖

WARNING
Lethal voltages are present. Exercise extreme caution when constructing and testing the control unit and never leave it upside down when children are present.

TEST EQUIPMENT USED
H-P distortion analyzer, 331A
Heathkit sine-wave, square-wave, audio generator, IG-5218
DMM, Radio Shack 22-168A
Oscilloscope, Proteck, Model 6502

PARTS SUPPLIER
RS = Radio Shack
RS dot Com = 800-442-7221
AES = Antique Electronic Supply 480-820-5411
ALEL = All Electronics 800-826-5432
Mouser = 800-346-6873

A 6AS7/6080 Circlotron Amp

Searching for a better sound from your speakers? Try this 6AS7/6080-based amp design featuring a circlotron configuration. **By Monny Nisel**

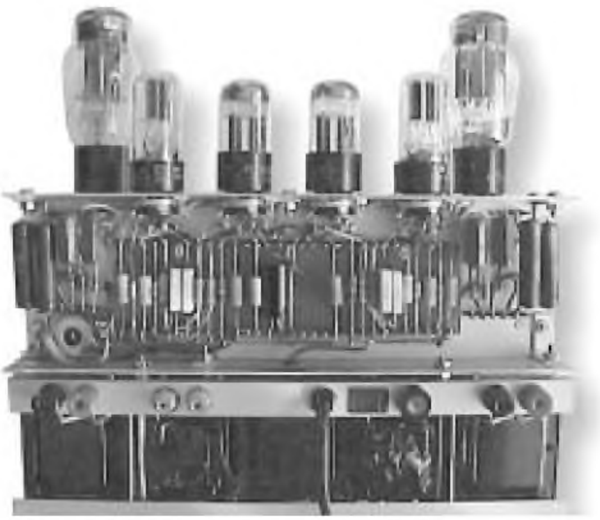


PHOTO 1: Rear view of 6AS7/6080 amp.

I started this project a couple of years ago. I owned a pair of moderately efficient speakers and wanted to build—on a limited budget—a good-sounding amplifier to drive them. So it was no surprise that I'd have to compromise somewhere. My search for a suitable schematic turned out ones with heavy and expensive output transformers or/and expensive output tubes. This was not exactly what I was looking for, so I decided to put something together myself. I ended up with a circlotron configuration using a single 6AS7/6080 with an output transformer, capable of delivering no more than 6 – 9W of output power but sounding beautiful.

I would like to stress from the beginning that this project, as conventional as it looks, is quite unorthodox with respect to some parts used and even some operating conditions of the output stage. It may not please the high-end sophisticated user, but may appeal to the hobbyist with a desire to experiment for a better sound. A look at the amplifier schematic (*Fig. 2*) shows a marriage between a Williamson-type input/splitter stage and a circlotron-type output power amplifier with a driver stage to interface the two. A power supply section of the schematic looking more elaborate than even the regular circlotron type should prove simpler and less expensive to build than it looks.

ABOUT THE AUTHOR

Monny Nisel is currently working part-time as a consulting electrical engineer after retiring from the aerospace industry in Montreal, Canada. He started building tube amplifiers in the early fifties and restarted designing and building in the nineties. He holds an electrical engineering degree from The Polytechnical Institute in Iassy-Romania and is a member of the Order of Engineers of Quebec Canada.

THE CIRCLOTRON

The circlotron output stage was patented in 1954 by Alpha M. Wiggins from Electro-Voice as a high-quality audio amplifier, and the company built several models in the following years. A simplified schematic of the circlotron configuration is shown in *Fig. 1*. The two output tubes are connected in series with two power supplies in a closed circle with the load connected to the cathodes of the tubes. In a theoretically balanced state, when the two currents (I1 and I2) through each half of the circuit are equal, they cancel each other through the load, so you must assume that the current flows only along the circle, hence the circuit's name.

Any imbalance between the two currents, such as the one generated by drive signals applied to the grids, translates in a current through the load following these signals. Driving the tubes with opposite phase signals the two currents in the load to add the same way as in a conventional push-pull circuit. But unlike in the classic push-pull circuit, the load is connected in parallel to the tubes and does not carry the DC supply, which is a great advantage.

There are a few more advantages, but the greatest and the one that prompted me to try this circuit was the unpretentiousness of the output transformer. When compared to the conventional push-pull amplifier, the impedance of the primary winding of a circlotron output transformer is just one-fourth of the plate-to-plate winding impedance of the former. A glance at the two configurations shows that for the same operating conditions and from the AC signal point of view, in the regular push pull you can regard each output tube as being in series with the entire (double) winding, whereas in the circlotron the two output tubes are in parallel with one half of that winding.

A low impedance output transformer means fewer turns in the windings as well as less distributed capacity and leakage inductance, hence a better frequency response. Switching transients are practically nonexistent because the current in the primary winding is never switched off regardless of the class of operation; there is no need for air gap in the core because no DC flows through the winding. So you deal practically with a simple, low impedance,

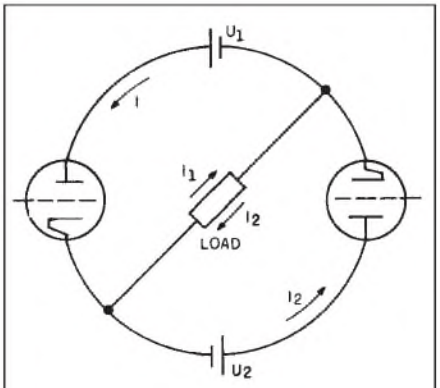


FIGURE 1: Simplified circlotron configuration.

single primary winding type transformer, operating in a push-pull mode due to an ingenious arrangement. Such a transformer costs a lot less than a regular push-pull transformer, and these savings may justify the cost of a more expensive power supply.

THE OUTPUT STAGE

One of the reasons for choosing the 6AS7 tube was my idea that I could eventually supply it with low plate voltage and high current, though it seemed to go against RCA recommendations as well as *GA*'s published projects using this tube. I wanted to try to push the current as high as possible to see what difference it would make in the sound.

Choosing a plate voltage of 120V, I plotted a 1kΩ load line in a single-mode operation and figured out a 1.8kΩ plate-to-plate load in a regular push-pull operation, or a 450Ω load impedance (one-quarter of 1.8kΩ) in a circlotron arrangement. It is common knowledge that a 450Ω impedance output transformer is not an off-the-shelf item, so I had to either have one custom-made or improvise something . . . at least to experiment with temporarily. The first choice would have been quite expensive and practically defeat the purpose of the full exercise, so I went straight to the second.

I figured out that an impedance ratio of 450 to 8Ω or 56/1 can be easily found in a power-supply-type transformer. A 120/16V transformer has a 7.5/1 turns ratio (120:16) or a 56.25/1 impedance ratio, which was exactly what I needed. Combined with the knowledge that the transformer could be a very simple type, I decided to try it and I was not disappointed; as a matter of fact the sound was so good I never had to revert to the first choice. The amp schematic is shown in *Fig. 2*.

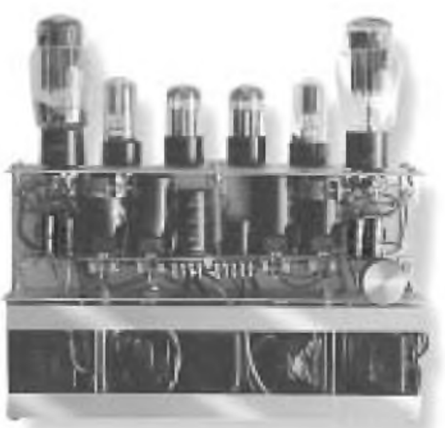


PHOTO 2: Front view.

With a 300Ω resistor in the cathode of each triode, the bias was close to 34V and current reached 110mA. I used the two trimpots P2 and P3 to balance the currents in the two halves of the tube. I found that most of the 6AS7s or 6080s I tried had slightly different currents when operating under the same condition, but sometimes the difference was quite noticeable.

SETUP

Because I wanted to avoid any DC current through the output transformer, I needed to balance the currents through both tubes. I made no attempt to adjust the current before the tube had reached thermal stability—approximately five to ten minutes after the amplifier had been switched on.

For those who noticed the somewhat unusual way of connecting the lower side of the adjusting potentiometers P2, P3 to the negative rail of the opposite tube, I found that in this way adjustment is a lot easier and more stable. As a matter of fact, the negative rail of the opposite tube is practically the negative rail of the same power supply whose positive one is connected to the tube for which you make the adjustment. With this setup the output power before clipping was close to 6W, provided the DC voltage supply of each tube could be maintained at 148–150V (*Table 1*).

My main supply voltage sometimes drops to 107V and the DC supply and output power follow suit; although it does not make a difference in volume or sound quality, it shows when I measure the power. At this low DC voltage supply even a few volts less in the main makes a difference in the output power. Because the two power supplies are floating with respect to ground, I connected the two 16kΩ resistors across the output transformer primary with the median point connected to ground, thereby producing an artificial reference to ground of the output stage.

The 16kΩ value, which may seem a little high for this purpose, is a compromise that should not short any significant amount of output power. This problem was solved in Wiggins' original design by using an output transformer with a middle tap connected to ground; the same can be accomplished here by using a 2 × 120/2 × 16V transformer, such as part #037013 201 from Plitron with the two windings on each side in series and the primary tap grounded. I noticed that some of the 6AS7s I used—especially the GA version—have a tendency to thermal runaway. I had better results with the 6080, especially the WA or WB versions or even with the 6AS7G, all having a bigger heat radiator strapped to the grid pylons; you should consider using this type of tube, especially when deciding to push the dissipated power a bit higher.

THE POWER SUPPLY

The power supply must deliver three different DC voltages and the 6.3V heaters supply. This task, as complicated as it looks at first glance, is accomplished with only two inexpensive off-the-shelf transformers. You can obtain the dual voltage for the output tube from a Hammond type 182 H 117 transformer, which has two separate secondary windings capable of delivering 117V at 342mA each. A bridge rectifier and a 2000μF capacitor complete the power supply for the output tube, delivering approximately 150V DC.

I found that for the output stage a single big capacitor is quite adequate instead of a full-blown pi filter using smaller capacitors. The perfect symmetry of the output stage leads to ripple cancellation in the load, to a degree that is barely audible even close to the speakers. The high voltage for the input/splitter and driver stages as well as the 6.3 AC for the heaters is supplied by a part #P-T6415 power transformer from Antique Electronic Supply.

I was concerned that the 6AS7 might fail during testing and short the power supply, so I inserted a 1A fuse in series with the positive rail. I never had a tube failure, but the fuse proved useful when I needed to disconnect one side or the other for testing and measurements.

All the electrolytic capacitors are computer grade, and I purchased them from surplus stores at a very low price. I found Mouser Electronics stores had a good selection of them. The same goes for the transformers; you could use any type with similar parameters.

In one version of the amplifier, I used a couple of 20VA 120/120V transformers, which I removed from bathroom-type isolation outlets, for the output tube power supply. The power supply is the most expensive part of the amplifier and reducing its cost greatly impacts the cost of the entire amplifier. In a stereo version of the amplifier you can replace T1 with part #P-T272JX power transformer (also from AES).

SPLITTER AND DRIVER STAGES

After experimenting with a couple of different designs, I settled for the Williamson type input/splitter stage. Though the cathodine splitter stage has less gain than other configurations, the input stage compensates for it, and apart from sounding better than other designs I tried, the entire stage uses one or two capacitors less than the other classical designs, a feature I liked. The following stage, the driver, had to provide over 200V pp to drive the output tubes biased at 34 to 59V DC, and for this purpose I had to supply the tube with higher voltage than the output stage, further complicating the power supply.

I used the same tube, the 6SN7, for both input/splitter and driver stages in an arrangement which is commonly used by manufacturers and experimenters alike. The 6SN7 is a very linear and good-sounding tube; it is also not expensive and can be easily found at any tube dealer and certainly in the possession of any respectable tube hobbyist. I used a RCA 6SN7 GTB tube for the driver stage and a Rogers tube as the input/splitter. I found that this combination has a good tonal quality, though I could not find significant differences using other brands of tubes.

I tried to keep the stage simple by not using trimpots to balance the output of the two sides. Instead, I chose to measure the driver's plate resistors to get their value as close as possible and have the tube with the two halves match. Apart from degenerative local feedback in the two stages, there is no global feedback; any small amount I tried was, in my opinion, muffling the sound, so I decided not to use it. The input voltage needed to drive the amplifier to full power was 350mV.

ASSEMBLY AND ADJUSTMENT

I assembled the amplifier in a stereo version in a somewhat unusual com-

PHOTO 3: Overhead view.

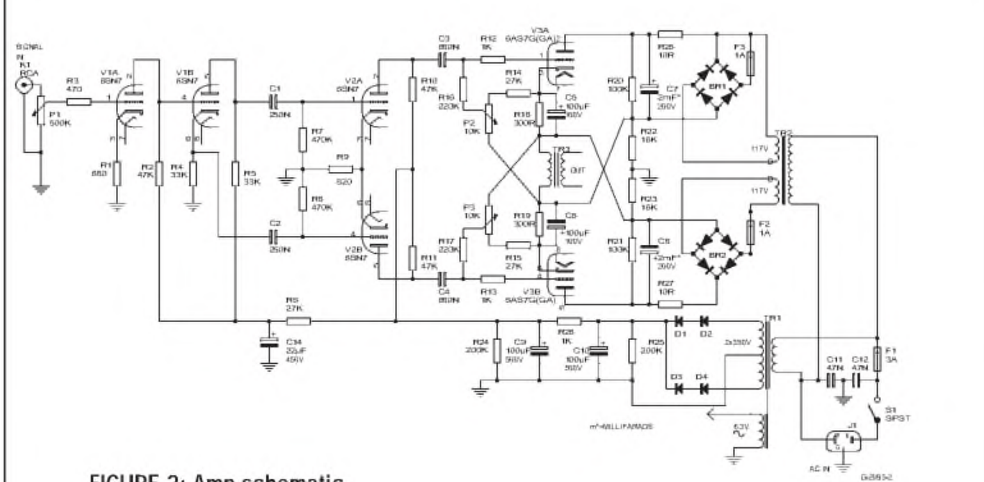


FIGURE 2: Amp schematic.

pact vertical chassis to fit a certain space in my system. Though this design has some advantages, I do not recommend it because of the difficulties of the mechanical work involved. Instead, you can easily install the components on a regular chassis, in a classic arrangement with all the transformers, tubes, and bigger electrolytic capacitors installed on top of the chassis and the smaller components underneath either in separate mono or stereo versions to fit your taste.

Once assembled and with components connected, the amplifier should work at once. However, you should check the correct connection of the components before power-up, especially at the output tubes where a mistake in biasing can overheat and eventually destroy the tubes if left unattended for a long time. You should take special care when wiring the trimpots P2 and P3 and the grid discharge resistors.

The next step is to adjust the output tube bias using P2 and P3 potentiometers. With a voltmeter connected in parallel to the cathode resistor R18, I adjusted P2 until I read 34V on R18, and repeated the same operation on the other half of the tube. With this bias voltage the current should not be more than 110 to 115mA. I made the adjustment only five to ten minutes after switching on the amplifier to let the tube reach thermal stability.

As mentioned previously, one of the reasons for using the 6AS7 was to find out how a tube already supplied with low voltage and high current will behave when current is pushed to the maximum—or even higher—recommended by the manufacturer. By turning P2 and P3 cursor to the cathode side, I could increase the current to over 140mA, which is above the 125mA maximum recommended. I always did it in small increments, leaving the tube to stabilize after each adjustment and watching for red spots on the plates.

At 140mA the voltage drop on the cathode resistor reached 42V, leaving the plate supply no more than 110V and sometimes less depending on the main supply. As the plate voltage dropped, so did the output power—as expected—and clipping settled in earlier. I could not measure distortions but they must have increased as well.

THE SOUND

Though the sound was quite nice from the beginning, once I started increasing the current the sound seemed to gain more body and become richer, especially in the lower side of the frequency spectrum, which made it really pleasant to listen to. At this point the bias was approximately 23V and the dissipated power reached 15W, way up from the 13W maximum recommended. Amazingly enough, I've never experienced any tube failure and I noticed no red spots on the plates.

As a matter of fact, a couple of 6AS7Gs I've been running for almost two years at close to 14W dissipation power in the amplifier I built seem to be working fine. Although in my experiments I went even further reaching 150mA plate current, I found there is no advantage in going that far. Apart from thermally overstressing the tube to the point of damaging it, power decreases dramatically and the distortion becomes audible even at lower volume.

So what is a good point to run the tube to take advantage of the higher current without going overboard? *Table 1* shows the parameters when I run the tube in four different conditions, with different plate voltages. I preferred to run the tube in condition no. 1 and increase the current to 125mA by adjusting R2 and R3 until the voltage drop on the cathode resistors reached 37V; the dissipated power is just above 13W. For those with 120V main, condition no. 2 gives close results.

For slightly increased power but also a little change in sound conditions, nos. 3 and 4 apply. Those two conditions in *Table 1* show that you can achieve higher output power by using higher supply voltage. I chose the 150 and 175V AC supply, respectively, because I found transformers with double secondary .2 × 150V and 2 × 175V at very reasonable prices. The 2 × 175V transformer is part #037053201 by Plitron and the 2 × 150V is part #RC 0100 040 1 from SUM R. In case you use the 175V supply, the electrolytic capacitors C7 and C8 should be at 250V minimum.

With regard to the output transformer, I experimented with two different types. I started with a regular E&I type 120/17V and switched over to a toroidal type 120/16V, both ranging from 80–100VA. Though there was not much of a difference in power output, each transformer had a different sonic signature, and though I preferred the sound of the latter, I used the former to build the amplifier due to assembly constraints.

TESTS AND MEASUREMENTS

I have not done any specific performance tests apart from output power and a frequency sweep at nominal

TABLE 1

		CONDITION #			
		1	2	3	4
Uac NL	volt	115	124	150	175
Uac FL	volt	113	121	144	169
Udc	volt	148	156	166	216
Ua	volt	112	120	126	156
Rk	ohm	300	330	375	700
Ug1	volt	–33	–36	–40	–59
Ia	mA	110	120	110	85
Pd	watt	12.3	14.4	13.8	13.2
Pu	watt	5.6	6.5	8	9

Abbreviations
Uac NL, FL: TR 2 secondary voltage with no load or full load
Udc: total rectified DC voltage
Ua: DC voltage between plate and cathode of the output tube
Rk: cathode resistor
Ia: plate current
Pd: dissipated power
Pu: output power

PARTS LIST

COMPONENTS	DESCRIPTION	REMARKS
RESISTORS		
R1	680R 1W	carbon composition (cc)
R2, R10, R11	47k 2W	metal oxide (mo)
R3	470R .5W	cc
R4, R5	33k 2W	mo
R6	27k 2W	cc
R7, R8	470k ½W	cc
R9	820R 2W	cc
R12, R13	1k 1W	cc
R14, R15	27k 1W	cc
R16, R17	220k ½W	cc
R18, R19	300R 10W	wirewound
R20, R21	100k 1W	cc
R22, R23	16k ½W	cc
R24, R25	200k 1W	cc
R26, R27	10R 2W	cc
R28	1k 2W	cc
CAPACITORS		
C1, C2	250nF 400V	polyester
C3, C4	680nF 400V	polyester
C5, C6	100µF 100V	electrolytic
C7, C8	2000µF 200V (250V)	electrolytic
C9, C10	100µF 500V	electrolytic
C11, C12	47nF 600V	polypropylene
C14	22µF 450V	electrolytic
RECTIFIERS		
D1–D4	1N4007	diode 1A 400 PIV
BR1, BR2	P-QBR-84 (AES)	bridge rectifier 8A 400V
TRANSFORMERS		
TR1	167P16 9(Hammond) 037013201 (Plitron)	117/16V-80VA or 115×2/15×2-80VA
TR2	182H117 (Hammond)	2×117V 342mA
TR3	P-T 64159 (AES)	700VCT 70mA 6.3V 3.5A
TUBES		
T1, T2	6SN7 GTA (GTB)	
T3	6AS7G,(GA), 6080	

power, due to a lack of dedicated equipment. However, I did extensive listening tests myself, and with other people, including a couple of audiophiles, and everybody liked what they heard very much; single instruments such as the piano or the guitar as well as female voices were really amazing. Sometimes it was difficult to realize that the sound came from an amplifier using this type of an "output" transformer. Any listening test should be done only after the amplifier has enough time to warm up (15 to 20 minutes).

I used three different types of speakers for listening tests: a pair of Fisher XP-1as, a pair of Acoustic Research AR 4xs, and a pair of JBL CF 120s. The best match was the JBL pair, which happens to be the most efficient, though with the AR 4xs, which are known for their low efficiency, the sound was strong enough to use no more than one-third of the volume in normal living room listening conditions. The frequency response of the amplifier seems to be very good from the listening test point of view, though the sweep showed a difference of ±3dB between 40 and 20,000Hz. The frequency response will be different for different transformers, and it seemed quite funny to measure the frequency response of power transformers meant to work only at 60Hz; amazingly enough some of the transformers measured have quite a wide frequency range.

I am not advocating the use of power transformers instead of normal output transformers, though I tried them on many other occasions with good listening results. I am convinced that a properly designed output transformer–or even an autotransformer, for that matter–would yield better performance. As I mentioned before, I tried to keep the cost as low as possible, making convenient compromises. Regarding the transformers for the output stage or power supply, you can use any type or manufacturer you find convenient, provided they have the same parameters as the ones specified.

CONCLUSIONS

This project includes a few features that make it different from other designs. First of all, it is a beginner's circlotron. Though more complicated than a regular amplifier, it is still affordable and can be assembled without great difficulty by anyone with little experience with tube amplifiers. Also:

- It uses a single 6AS7 in a circlotron configuration with an output transformer.
- It uses lower than recommended voltage and maximum current.
- The output transformer is a power transformer conveniently chosen to match the output impedance of the stage.
- The cost is kept as low as possible by using inexpensive—and where possible—surplus parts.

AND A FEW MORE WORDS

The experiment bug drove me to break another taboo. I tried the amplifier with fixed bias without any cathode resistors and with 145V DC at the plates, and it seemed to work quite well. The output was 9W, but, again, it produced a different sound that I liked less than the one from the automatic bias version. ❖

SOURCES

Antique Electronic Supply
6221 S Maple Ave
Tempe AZ 85283
480-820-5411
FAX 480-820-4643
info@tubesandmore.com
www.tubesandmore.com

Hammond
256 Sonwil Dr.
Cheektowaga, NY 14225
716-651-0086
FAX 716-651-0726
www.hammondmfg.com

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

Plitron
8-601 Magnetic Drive
Toronto, ON M3J 3J2
Canada
416-667-9914
FAX 416-667-8928
1-800-PLITRON (1-800-754-8766)
www.plitron.com

SUM R, Richard Sumner Technology Inc.
29 Connell Court, Unit #17
Etobicoke, Ont., Canada M8Z 5T7
(416) 251 2523
FAX (416) 251 0401
www.sumr.com

Make a Panel-Cutting Jig

With the right table-saw setup, you can perform woodworking tasks with much greater ease. **By Bill Fitzmaurice**

Over the years I've heard from quite a few readers regarding power tools—most ask how they can build top-quality projects without spending thousands of dollars on tools. A recent inquiry, in particular, wanted to know how to build a DR12 horn in the absence of an eight-hundred-dollar table saw, perhaps by having a lumber yard pre-cut the parts. The problem with that scenario is that my projects involve a lot of "trim-to-fit" steps, so pre-cut parts are not a viable option. The good news is that I do not own an eight-hundred-dollar table saw either, and that you can build just about any cabinetry with a two-to-three-hundred-dollar table saw, or even better, buy a used one for a hundred bucks or so.

This will understandably shock those of you who regularly peruse tool catalogs, knowing that you can easily spend five hundred dollars on a rip fence alone, let alone a saw on which to mount it. But the truth is that even a mediocre saw can turn out perfectly square and true panels of just about any size as long as you have a panel-cutting jig. My old trusty jig, after 15 years use and hundreds of projects, needed replacing recently, so I took pictures of the process to show you how you can make what I consider to be the most important piece of equipment in my shop.

JIG CONSTRUCTION

The first step is cutting a piece of plywood to serve as the jig. I used a piece of ¾″ Baltic birch, for its inherent stability, cut to about 30″ × 40″ inches. Don't go to any great lengths to make this piece perfectly square—it doesn't matter. Alternately, you may opt for ¾″ MDF, and for a fancy look maybe even top that MDF with Formica.

The advantage of Baltic birch is that you can cut the runners you'll need from it as well. The runners fit into the two slots that run across the top of your saw table; the standard slot size is ¾″ wide, 3/8″ deep. I wouldn't make the runners from standard five- or six-ply plywood, but ten- or eleven-ply Baltic birch works well. Alternately, you could make the runners from solid wood, preferably a hardwood such as oak.

In any event, cut the runners 3/8″ thick, a few inches longer than the length of your jig (*Photo 1*). Another good reason to use Baltic birch is that nominal ¾″ plywood is about 1/64″ thinner than advertised, so the runners will slide freely through the slots on the table. If you use ¾″ stock, you'll probably need to sand them a bit.

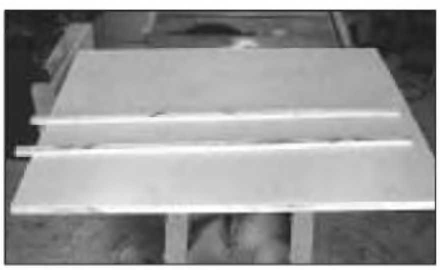


PHOTO 1: One large piece of plywood and two small strips form the basis of the panel-cutting jig.

PHOTO 2: Runners in place in the table saw slots.

When you have your runners cut and sanded (if necessary), place them in the slots (*Photo 2*). Lower the saw blade fully below the table. Run beads of wood glue on top of the runners, place the plywood atop them, and drive 1″ brads through the plywood into the runners (*Photo 3*). The plywood is centered on the saw blade; in the case of a 40″ wide jig, that leaves 20″ to either side of the blade.

Remove the jig and raise the blade fully. Start the saw. Place the ends of the runners (these extend a few inches past the jig's leading edge so that you can easily align them with the slots) and slowly push the jig across the table,

PHOTO 3: Driving brads through the jig into the runners.

PHOTO 4: Cutting the jig almost, but not quite, in half.

cutting through the jig to within about 2″ of the back end of the jig (*Photo 4*).

PUSH-BAR

Next comes the push-bar, which must be made of hardwood; I made mine from ¾″ oak, 4″ wide by 30″ long. Mount this at the rear of the jig, screwed and glued to the plywood. Determine its mounting by placing a framing square along the saw blade to derive a perfect right angle to the blade, and clamp the push-bar to the jig so aligned (*Photo 5*). Use the clamps to hold the parts in alignment and drive at least six screws from the jig underside into the push-bar, pre-drilling and countersinking.

If you happen to have a piece of hardwood at least 2″ thick for the push-bar, you're done. If not, you'll need to laminate some more stock to the back of the push-bar to stiffen it further (*Photo 6*). Note that the stiffener need not extend the full height of the push-bar, as you end up cutting the lowermost inch or so in the final step, anyway. That final step is to run the jig fully across the blade, cutting partly through the push-bar (*Photo 7*).

USING THE JIG

Using the panel-cutting jig is the utmost in simplicity. Put the part you wish to cut on the jig, one edge firmly against the push-bar, and push the jig across the saw table. The resulting cut will always be at a perfect 90° angle.

You may use the jig for cutting panels as large as 36″ × 48″ or so. If you need to cut larger panels, just make a bigger jig. But don't reserve it for large parts only; you'll find yourself using the jig any time you need a right-angle cut, no matter how small the piece.

One benefit of the jig is that it allows you to easily square all four sides of a panel, which is critical for fine work. Plywood these days seldom comes from the factory perfectly squared; four passes atop the panel-cutting jig will give you better-than-factory accuracy. Another benefit is safety.

Those of you who are astute—or perhaps litigious, will note that there is no blade guard on my saw. In fact, you will seldom find a blade guard on any professional's table saw. Accidents happen when digits and limbs come into contact with blades, and the best way to prevent that is to never handle the piece of wood that you are cutting. Pros always use hold-downs, push-bars, feather-boards, and jigs so that hands never come near blades.

With the panel-cutting jig, you push the jig across the saw, not the work-piece. Working with small parts, you may clamp the work-piece to the push bar. You may make perfectly true angled cuts by screwing the work-piece to the jig across the cut line at the correct angle, holding it firmly in place while being cut. You may even cut and square large assemblies with ease (*Photo 8*).

The best part of this project is the cost, easily less than $30. You will achieve accurate cuts better than with even the most expensive rip-fence. You'll also increase the safety factor of your projects, and you'll be amazed how much faster the measuring and cutting process is compared to using a rip-fence. Precision work does require precision tools, but not necessarily a lot of money to acquire them. ❖

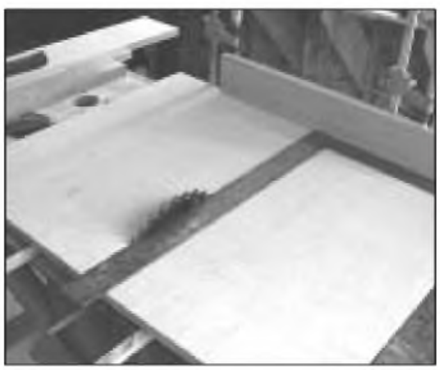


PHOTO 5: Squaring the push-bar to the blade.

PHOTO 6: Reinforcing the push-bar.

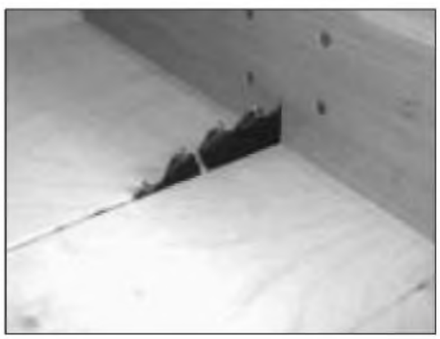


PHOTO 7: The blade passes through the lowermost portion of the push-bar.

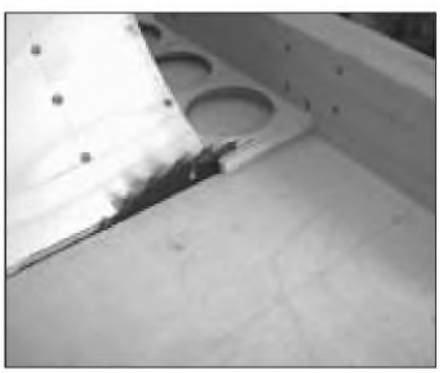


PHOTO 8: Trimming and squaring with the panel-cutting jig.

Product Review

Budget Phono Preamps

Reviewed by Charles Hansen and John and Sandra Schubel

If you wish to archive your vinyl records on CD-Rs, or you have a newer receiver or preamplifier that does not have a phono preamp built into it and wish to connect a turntable, you need an outboard phono preamp.[1,2] This is a test of some budget stand-alone moving magnet (MM) phono preamps.

Many of the units in this article won't be found at your high-end audio dealer. Think Best Buy, Circuit City, K-Mart, and The Wiz. They are also available on the Internet (eBay always seems to have several of them) and at catalog companies such as Crutchfield, MCM, Parts Express, and Radio Shack. Two are available from kit suppliers.

So, if you have thought of purchasing one of these preamps, what can you expect in terms of performance?

First, there are very few specifications with some of these preamps. The input resistance—and capacitance (if any)—is fixed in these preamps. I chose seven different units for my testing. *Table 1* also includes the specs for the highly regarded discrete phono section of the NAD 304 integrated amplifier.

Hagerman Bugle (kit, www.hagtech.com)	$50
MCM Model #40-630 (www.mcm.com)	$13.50
PAiA 9802K (kit, www.paia.com)	$23.75
Parts Express Rolls VP29 with AC adapter (www.partsexpress.com)	$77.99
Radio Shack 970-1018 (www.radioshack.com)	$24.99
Tech Link TPA2 (www.tracertek.com)	$55
TCC TC-400 with 12V DC adapter (www.tradertrax.com)	$26.95

In this age of op amps, three of the seven units have discrete transistor circuitry. None have regulated power supplies, just R-C filters. Some run on 9V batteries.

There were also several older Shure phono preamps for sale on eBay, but I wanted to get new, commercial units. You can explore this used-equipment market yourself, if you like. I also passed on a unit from Terra Tec (with editing software, www.terratec.net) that uses a D-connector to interface with a PC and obtains its power from the PC's soundcard game port. That eliminated its use in an audio system.

THE BUGLE

The Hagerman Bugle Audiophile Phono Stage is available as a $50 kit, or a $25 "half-kit," which includes a bare epoxy PC board and the assembly manual. You can buy the electronic components from DigiKey (www.digikey.com) yourself, or use your own stock of parts. An assembled version will also be available (check the website).

The 13-page manual is detailed and thorough. The finished kit has no chassis and uses two on-board 9V batteries for a power supply (*Photo 1*). Battery life is estimated for 16 hours. Parts quality is first-rate, with polypropylene caps, 1% metal film resistors, and gold-plated jacks.

A brass screw allows you to connect your turntable to the circuit ground to reduce noise. The kit components provide the standard 40dB RIAA equaliza-

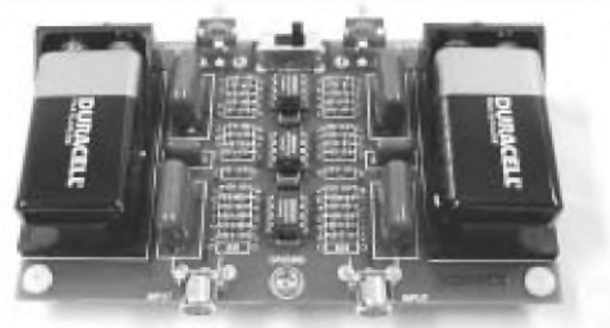


PHOTO 1: Hagerman Bugle board.


PHOTO 2: MCM Model #40-630 interior.

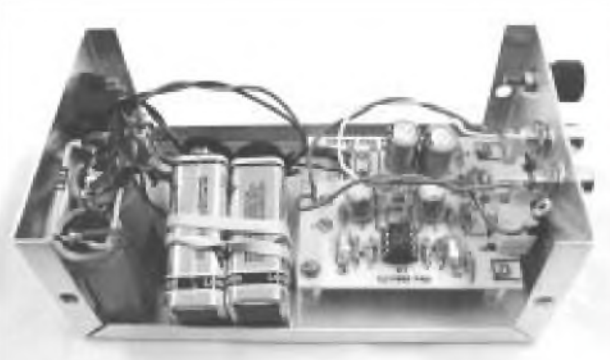


PHOTO 3: PAiA Model 9802k kit board in enclosure.

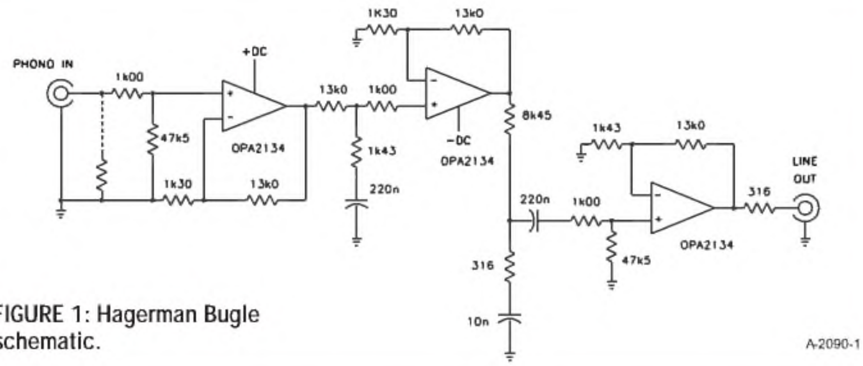


FIGURE 1: Hagerman Bugle schematic.

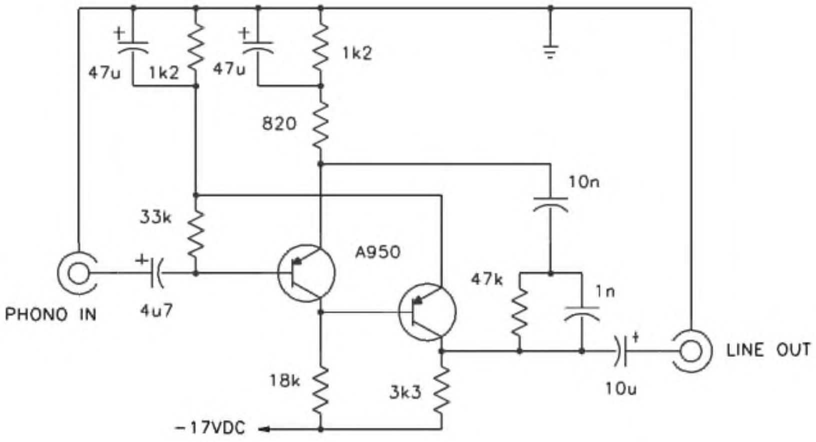


FIGURE 2: MCM Model #40-630, Bozak-Madisson CLK-PH2 schematic.

tion curve, but you can customize the Bugle for any EQ curve you desire, with 40dB to 60dB gain. An on-line component value calculator is available at www.anyeq.com.

The circuit schematic (*Fig. 1*) uses three OPA2134 dual high-performance audio op amps. The first stage boosts the magnetic phono cartridge signal by 21dB. An empty resistor position at each input allows you to provide lower resistance loading for moving-coil cartridges. The three-step RIAA curve is implemented in two passive EQ stages (all the other preamps use feedback, or active, EQ).

The RC network between U1 and U2 provides the 3180μs and 318μs breakpoints. The RC network after U2 provides the 75μs breakpoint. An additional series RC before U3 adds a low-frequency −3dB rolloff below 15Hz, and removes any DC offset passed from the first two stages.

MCM AND BOZAK-MADISSON PREAMPS

The MCM Model #40-630 solid-state and Bozak-Madisson CLK-PH2 stereo

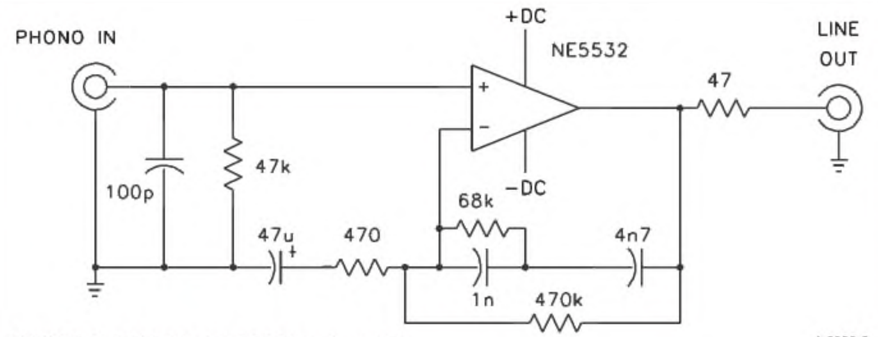


FIGURE 3: PAiA Model 9802k schematic.

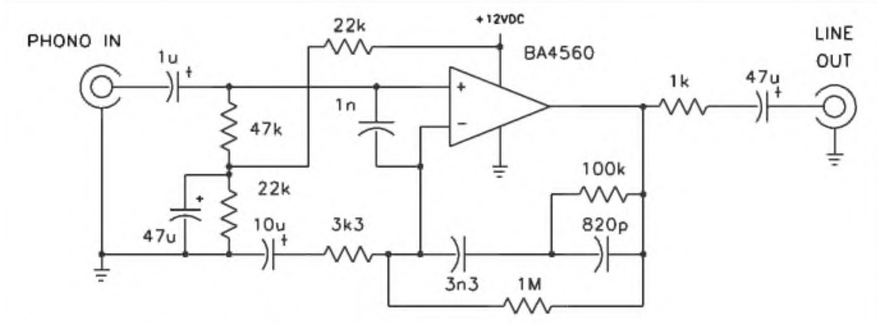


FIGURE 4: Parts Express #245-035 schematic.

phonograph preamplifiers are both available on eBay. These identical made-in-Taiwan units have a discrete circuit, with two 2SA950 PNP transistors per channel. The interior of the unit is shown in *Photo 2*.

The preamp has a painted steel chassis with a chem-filmed steel bottom plate. A plated steel shield sits between the power transformer and the epoxy circuit board. All four jacks are tin-plated. The phono input jacks are connected to the board via bare leads, while the line output jacks are soldered directly to the board.

The basic schematic is shown in *Fig. 2*. The half-wave rectified power supply uses an R-C pi-filter. The circuit board draws about 5.5mA of quiescent current.

The circuitry is similar to the phono preamps used in 1970s-vintage transistor stereo systems. All resistors are 5% carbon film, and the EQ capacitors are general-purpose (not quality NP0) ceramics. Cartridge loading is by a dis-

TABLE 1
SPECIFICATIONS FOR THE SEVEN PHONO PREAMPS

PARAMETER	NAD 304	HAGERMAN	MCM	PAiA	PARTS EXPRESS	RADIO SHACK	TECH LINK	TCC
Input Sensitivity, 1kHz	$10mV_{in}$ – $660mV_{out}$		$6mV_{in}$ – $500mV_{out}$			$5mV_{in}$ – $450mV_{out}$	$2.5mV_{in}$ – $150mV_{out}$	$5mV_{in}$ – $450mV_{out}$
Input Overload,								
20Hz	22mV							
1kHz	220mV	55mV	30mV			38mV max		
20kHz	2V RMS							
Gain, 1kHz	30dB	40dB				34dB	30dB	
Input R,C	47k0. 200pF	47k, 14pF	50k			50k	50k, 1nF	20k
Response		15Hz–150kHz	30Hz–20kHz	"RIAA"		30Hz-–20kHz	"RIAA"	18Hz–18kHz
RIAA Accuracy	±0.5dB	±0.5dB				±1dB		+5dB, –1dB
Output (10k load)	N/A		1.8V max			1.8V RMS	1.5V RMS	
Output Impedance		330Ω				50k		500Ω
Distortion	<0.1% at +30dB	0.05%, 1kHz					0.004%	
S/N. A Wtd, Ref 5mV	77dB	74dB	60dB			50dB	80dB	50dB
Crosstalk, 10kHz						>50dB	–65dB	

TABLE 2
MEASUREMENTS FOR THE SEVEN PHONO PREAMPS

PARAMETER	HAGERMAN	MCM	PAiA	PARTS EXPRESS	RADIO SHACK	TECH LINK	TCC
Input Sensitivity, 1kHz	$10mV_{in}$ – $1.01V_{out}$	$10mV_{in}$ – $610mV_{out}$	$10mV_{in}$ – $1.57V_{out}$	$10mV_{in}$ – $306mV_{out}$	$10mV_{in}$ – $480mV_{out}$	$10mV_{in}$ – $380mV_{out}$	$10mV_{in}$ – $1.35V_{out}$
Input Overload,							
20Hz	7.4mV	6.3mV	10mV	19mV	2.4mV	4.7mV	3.3mV
1kHz	57mV	53mV	58mV	152mV	23.6mV	50mV	11.5mV
20kHz	510mV	255mV	440mV (see text)	1.35V	94mV	58mV	63mV
Gain, 1kHz	40dB	35.7dB	43.9dB	29.7dB	33.7dB	31.6dB	42.6dB
Zin, 1kHz	48k5	32k	46k7	47k4	26.5k	24k	152k
RIAA Accuracy 30Hz–20kHz	±0.6dB	+5, –3.8dB	+2, –3.9dB	+0.9, –1.8dB	+0, –0.89dB	+13, –0.3dB (NAB curve)	+3.6, –0.5dB
V_{out} (10k load, 1kHz, 1%THD)	5.6V RMS	3.1V RMS	8.9V RMS	4.6V RMS	1.12V RMS	1.75V RMS	1.46V RMS
Output Impedance	307Ω	1k02	48Ω	970Ω	450Ω	890Ω	995Ω
Distortion, 1kHz, 10mV in	0.03%	2.1% (see text)	0.008%	0.008%	0.18%	0.13%	0.30%
S/N, A Wtd, Ref 10mV	72dB	33dB	75dB	86dB	54dB	68dB	60dB
Crosstalk, 10kHz	–62dB	–43dB	–43dB	–64dB	–58dB	–46dB	–62dB

PHOTO 4: Parts Express #245-035 interior.

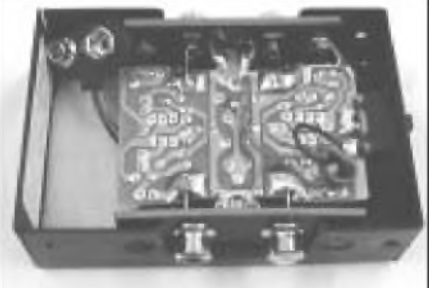


PHOTO 5: Radio Shack 970-1018 interior.

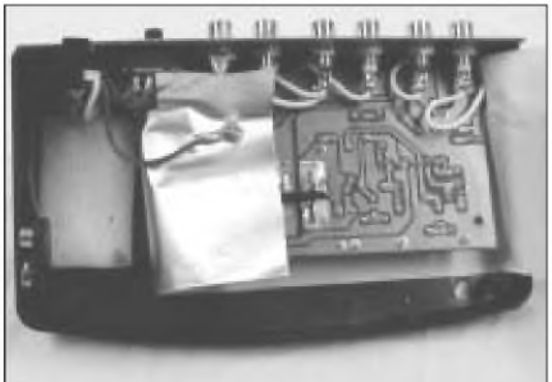


PHOTO 6: Tech Link TPA2 interior.

crete 33k resistor and the transistor circuit impedance.

The pricier Bozak-Madisson version of this preamp comes with a useful 19-page tutorial on transferring vinyl to CD-R using a CD recorder or PC, and an explanation of how to use Ganymede's Wave Corrector software. It also comes with the RCA-to-3.5mm phone jack patch cord you need for connecting it to your computer sound card.

PAiA 9802K

The PAiA Model 9802 RIAA is a bare PC board kit without power supply or chassis. As with the Hagerman preamp, the advantages of a kit are that you can upgrade the components to whatever you like. The basic circuit topology is fairly decent (*Fig. 3*).

The standard components are based on the NE5532 dual low-noise opamp. The EQ caps are high-quality 5% polystyrenes, with 5% carbon film resistors used throughout. A 47k resistor and a 100pF X7R ceramic cap supply the cartridge loading. A pad is available on the PC board for a turntable ground connection.

The unit is designed for a power-supply range of ±9 to ±18V DC, and PAiA makes a line of regulated dual supplies in kit form. They recommend their ±12V DC model 9770R-12. You can also use two 9V DC batteries.

Two tin-plated RCA input jacks are provided for mounting on the single-sided phenolic PC board. Two ¼″ mono phone jacks are provided for the outputs. The chassis is up to you.

I decided to build the kit with a high-quality gold-plated 8-pin DIP socket, and I used gold-plated phono jacks

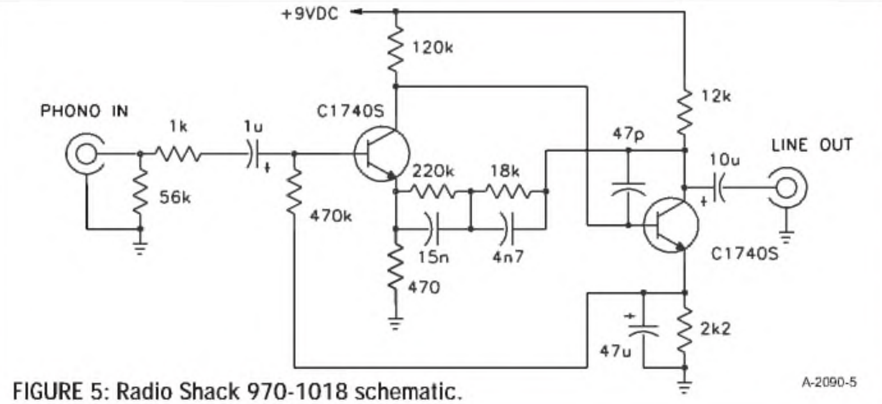


FIGURE 5: Radio Shack 970-1018 schematic.

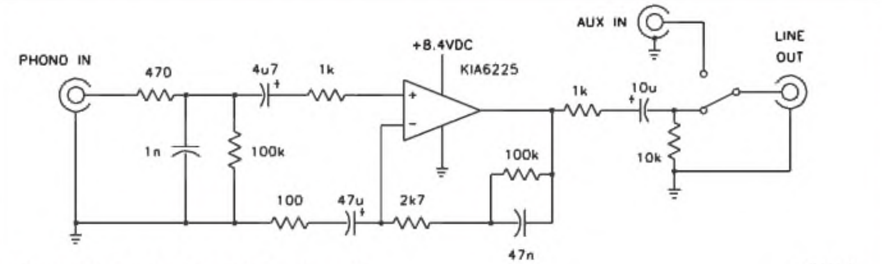


FIGURE 6: Tech Link TPA2 schematic.


PHOTO 7: TCC TC-400 interior.

(*Photo 3*). I made an unregulated ±15V DC, dual half-wave rectified and filtered power supply from a 9V AC plug-in power adapter[3]. I put a 47k bleeder resistor across the 2200µF filter caps. A DPDT-CO switch selects either the power adapter or two 9V batteries.

I used a RS 270-238 mini-box for the chassis, with a binding post to attach the turntable ground wire. Other than that, I used the parts supplied with the kit. The extra parts and the chassis added about $20 to the cost of the preamp.

PARTS EXPRESS PREAMP

The Parts Express #245-035 Rolls VP29 Phono Preamp (*Photo 4*) is made in Salt Lake City, and comes with a 12V DC power adapter. It has the biggest power-supply filter cap (1000µF) and a green LED to show the unit is active. The two-piece steel chassis is painted crackle-finish red. Gold-plating is used on the outer shells of the RCA input and output jacks, with tin center contacts. The preamp also has a gold-plated ¼″ stereo phone output jack, and the chassis has a binding post that connects to circuit ground via a 50nF cap, so you can connect your turntable ground lead to the preamp.

The circuitry (*Fig. 4*) is based on a BA4560 dual low-noise op amp. The op amp is socketed, so you can easily change it if you like. The EQ caps are Mylar film, with 5% carbon film resistors used throughout. A 47k resistor returned to half the supply voltage (phantom ground) supplies the cartridge loading.

RADIO SHACK PREAMP

The Radio Shack 970-1018 magnetic cartridge stereo preamp is another battery-powered preamp, made in Korea. It is not carried in the RS stores; the part number on the RS website was 970-1018, while the box I received was labeled catalog number 42-2111. It has a discrete circuit, with two 2SC1740S NPN transistors per channel (*Photo 5*).

The painted steel chassis has a flip-open brushed aluminum cover that is secured by two screws. The on-off switch and the four tin-plated RCA jacks are soldered directly to the PC board.

SOLVING FOR INSTABILITY

When John Schubel discovered the Parts Express Rolls VP29 caused his amplifier's protection circuit to trip, he requested a second sample. Unfortunately, this second unit had the same malady, so he invited me over to hear for myself. When the preamp volume was advanced to anything above polite dinner party background music level, there was an oscillation at very low frequency just prior to the trip. This correlated with the random ultra-low frequency DC offset wander (about ±0.1V) I noted during measurements, almost like a record warp.

I ran a SPICE simulation of the circuit, extending the AC analysis from 0.1Hz to 10MHz, to look for degraded stability that could trigger oscillation. I fed a SPICE sine-wave generator to the preamp through a simulated inverse RIAA network to reflect the rising input signal level with frequency. While the audio from an LP would certainly not rise indefinitely due to the cutting lathe velocity limit, surface noise might.

The VP29 uses a 1nF capacitor across the input terminals of its 4160 opamps. This differential mode capacitor (Cid) forms a response pole in conjunction with the RIAA feedback network. Cid causes additional phase shift, and produces gain peaking by partially shunting the feedback gain setting RC network.

In the SPICE simulation, the input signal continues to rise with frequency while the preamp gain levels off (flat RIAA response) from 10Hz to 30kHz. Above 30kHz the preamp gain increases again until it meets and tracks the input signal level from 476kHz to 730kHz. At the same time, the input and output signal phase angles are decreasing.

Suddenly, at the pole formed by Cid, the output sees a phase reversal from −180° to +180°, while the gain is still near unity. The phase margin drops to about 50°, setting the stage for oscillation. At this high frequency, the op amp is also trying to drive the very low impedance of the two series RIAA network caps, depriving its internal compensation capacitor of current in the process.

I changed the SPICE Cid from 1nF to 1pF, and the phase margin problem went away. Although the gain remained near unity above 700kHz, the output also remained essentially in phase with the input. While the trigger appears to be HF instability, the oscillation seems to get started at LF, based on what John and I heard just before his amp protection tripped, and the wandering DC offset I measured. I should also mention that none of the other preamps use a differential input capacitor across their op-amp inputs.

The schematic is shown in *Fig. 5.* The data sheet lists the quiescent current as 1mA. All resistors are 5% carbon film, and the two EQ capacitors are Mylar film. An NP0 ceramic is connected b-c at the output transistor. Cartridge loading is by a discrete 56k resistor and the circuit impedance.

TECH LINK PREAMP

The Tech Link TPA2 is an aux/phono preamplifier made in China with a plastic enclosure. In order to provide shielding, the epoxy PC board is wrapped in aluminum foil tape that has a paper/plastic insulating layer under it (*Photo 6*). A jumper lead connects the foil to the power jack ground. The unit can be powered by an internal 9V battery or a 9V DC plug-in adapter that you must supply (with a 2.5mm × 5.5mm plug with +9V on the center pin—nothing in the package tells you this).

The internal polarity protection diode drops this to about 8.6V DC, with quiescent power consumption from the 9V battery at about 3mA. Two of the four suction cup feet partially block the battery compartment and must be held out of the way to access it.

While some of the other units have RCA jacks with gold-plated shells and

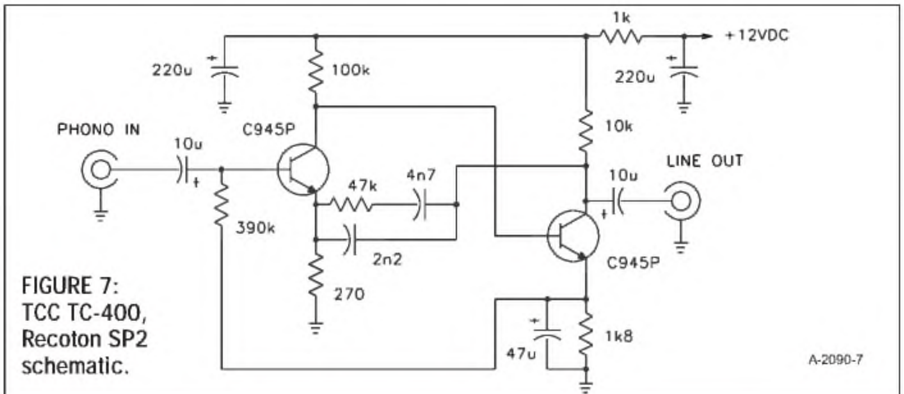


FIGURE 7: TCC TC-400, Recoton SP2 schematic.

tin center contacts (fool's gold?), the six RCA jacks in the TL preamp are fully gold-plated. A rocker switch on the cover selects either the AUX (directly switched to the output) or the phono input.

The preamp uses a KIA Electronics KIA 6225S dual preamp IC in a 9-pin SIP package. The schematic (*Fig. 6*) is taken from the KIA data sheet for an NAB tape preamp circuit (www.kecorp. com), with a change to the EQ capacitor value, so only one of the three RIAA EQ breakpoints is implemented! All resistors are 5% carbon film, and the lone EQ capacitor is Mylar film. Cartridge loading is the parallel combination of a 100k resistor, the 100k± 50k input impedance of the 6225, and a 1nF X7R ceramic cap.

TCC AND RECOTON PREAMPS

The TCC TC-400 audio stereo phono preamplifier and the Recoton SP2 stereo phono preamplifier are two more essentially identical phono preamps. The interior of the made-in-Taiwan TCC unit is shown in *Photo 7*. The black-painted steel enclosure is very similar to that of the MCM and Bozak-Madisson units, except it uses a 12V DC power adapter that is furnished with the unit. They also supply a set of generic tin-plated phono interconnects.

The cardboard box, with a Calrad Electronics p/n 80-574 sticker, has an Audio Precision frequency-response graph printed on the top. It shows the low-frequency (LF) response rising to +5dB at 20Hz, and the high-frequency (HF) response dropping to −1dB at 20kHz. According to the box, a version with both RCA and DIN jacks is also available as the TC-400D.

The input and output jacks have gold-plated shells, but tin center contacts. These jacks are connected to the PC board by two-pair shielded wires and Molex-style 3-pin connectors. There is no provision for a turntable ground connection.

The circuitry (*Fig. 7*) is similar to that of the Radio Shack unit, with a different EQ network topology. Two 2SC945P NPN transistors are used per channel. All resistors are 5% carbon film, and the two EQ capacitors are Mylar film.

MEASUREMENTS

For S/N and DC offset measurements, I terminated the preamplifier input jacks with a "cartridge" load consisting of a 499Ω metal film resistor mounted in a shielded phono plug. The line-level output load for all tests was 10k0.

I used an inverse RIAA network for frequency response and some distortion measurements. I used the distortion test set 80kHz low-pass filter to limit out-of-band noise during the distortion tests. I made response and distortion versus frequency measurements with a test signal level into the inverse RIAA network that produces 10mV at 1kHz at the preamp input jack. This is equivalent to a cartridge with an output of 10mV at 5cm/s recording velocity (2mV/cm/s sensitivity).

Typical vinyl records are recorded at 5cm/s maximum, while the RIAA specification allows a maximum recording velocity of 25cm/s (50mV at 1kHz for my testing). Therefore, any phono preamp that exceeds 1% THD+N with an input of 50mV at 1kHz would not meet the RIAA specification. You can see from *Table 1* that many of these preamps are specified for inputs of only 2.5mV to 6mV. Yet many of the "DJ" and some standard MM cartridges do indeed have 10mV outputs. This type of phono preamp will likely be employed in a live-music "DJ" setting.

I also measured the gain at 1kHz, for comparison with the RIAA standard of +40dB at 1kHz. The low power-supply voltage of some preamps may require a lower overall gain to prevent clipping at

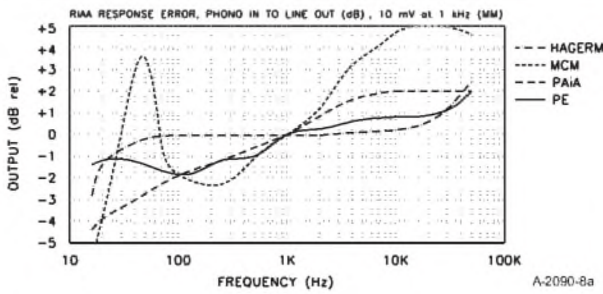


FIGURE 8A: Relative RIAA equalization error (Hagerman, MCM, PAiA, PE).

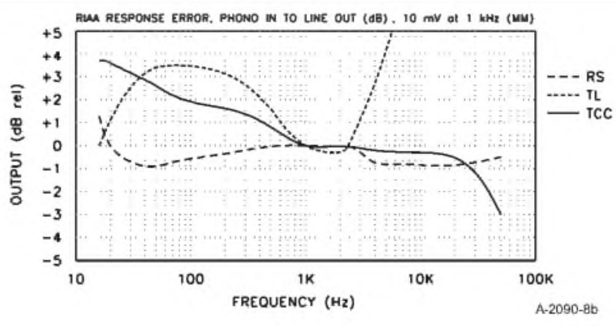


FIGURE 8B: Relative RIAA equalization error (RS, TL, TCC).

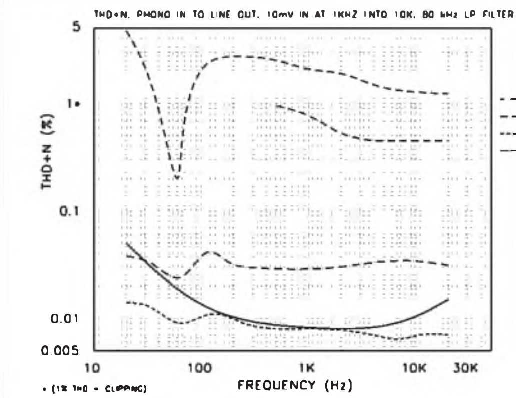


FIGURE 9A: THD+N versus frequency (2mV/cm/s) (Hagerman, MCM, PAiA, PE).

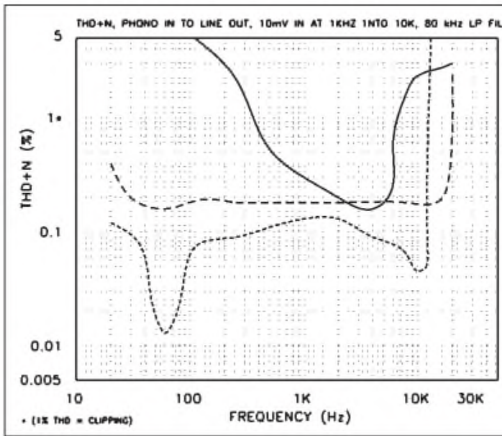


FIGURE 9B: THD+N versus frequency (2mV/cm/s) (RS, TL, TCC).

low frequencies where the RIAA gain requirement is highest. You will need to make up this gain shortfall in your line preamp or computer sound card.

My signal-noise and crosstalk 0dB reference level is the output that each card produces at 1kHz with a 10mV input. For distortion versus output at 1kHz, I fed the sine-wave generator directly into the phono preamp without the interposing inverse RIAA network. *Figures 8A* and *8B* show the relative RIAA equalization error, where 1kHz is the 0dB point. To get the true picture, you must add in the overall gain at 1kHz for each unit, which is listed in *Table 2*. Any designed-in LF rolloff is acceptable as long as it meets the 7950µs (20Hz) time constant[4] and the rest of the curve is flat.

Figures 9A and *9B* show THD+N versus frequency at a reference input level of 2mV/cm/s, while *Figs. 10A* and *10B* show THD+N versus output voltage at 1kHz. Again, the data is presented in two graphs for better clarity.

THE BUGLE

The Hagerman Bugle maintains normal output polarity. Input impedance was 48k5, and output impedance was 307Ω, all at 1kHz. The output noise was 0.25mV left and right (–72dB). The maximum DC offset voltage was +6mV (left).

Gain at 1kHz, 10mV input was exactly 40dB. The RIAA accuracy, as shown in *Fig. 8A*, was within ±0.6dB from 30Hz to 20kHz. It continues to rise at higher frequencies. Crosstalk at 10kHz measured a low –62dB in both directions.

The THD+N versus frequency (2mV/cm/s input level) is shown in *Fig. 9A*. Initially, I saw 0.12% THD at 1kHz (in the higher left channel). However, when I connected the inverse RIAA network chassis to the preamp ground screw, it dropped to 0.03%. The distortion residual showed low-level noise with no discernible harmonics.

The THD+N versus line output level at 1kHz is shown in *Fig. 10A*, with fresh 9V batteries. The input overload at 1% THD clipping was acceptable for a 10mV cartridge: 7.4mV at 20Hz, 57mV at 1kHz, and 510mV at 20kHz. Clipping was delayed by distributing the gain over three op-amp stages.

MCM PREAMP

The MCM Model #40–630 phono preamp maintains normal output polarity. Input impedance was a low 32k, and output impedance was 1.02k, all at 1kHz.

DC offset voltage was 30mV. The output noise, at 13mV (–33dB), is dominated by 60Hz hum, as is the distortion residual. The transformer inside the module, and a circuit topology that has a lower power-supply rejection capability than the op-amp-based designs, aggravate this problem. Interestingly, the 60Hz output noise dropped to 4mV with the input open-circuited. This suggests a ground-loop problem may also exist.

The instructions for this preamp specifically tell you not to connect the turntable ground lead to the preamp. In my own audio system, the hum was noticeable from my listening position with my preamp volume control at 12 o'clock. The preamp comes with a polarized two-prong AC line plug, so I did not try to reverse it with a "cheater."

Gain at 1kHz, 10mV input was 35.7dB. The RIAA accuracy was skewed towards the higher frequencies, with an LF peak at 48Hz as shown in *Fig. 8A*. I could also see some high-frequency oscillation on the falling sides of the sine wave when the test frequency exceeded 2kHz. Crosstalk at 10kHz measured –43dB in both directions (the test-set bandpass filter rejected the 60Hz hum).

The THD+N versus frequency (2mV/cm/s input level) is shown in *Fig. 9A*. I took one reading at exactly 60Hz, where it was 0.2%. This is because the fundamental notch filter removes the 60Hz hum as well as the oscillator signal.

I re-ran the test from 500Hz to 20kHz using the 400Hz high-pass filter in the distortion test set. With the 60Hz hum attenuated, the distortion performance is much better, as represented by the partial HF curve in the MCM data. Here the distortion residual consists mainly of even-order harmonics.

The THD+N versus line output level at 1kHz is shown in *Fig. 10A*. Again I took the data with and without the 400Hz HP filter. The data with the filter is the lower curve at 1V output. It isn't until the output voltage is almost 3V RMS that the signal is finally great enough to overcome the 60Hz hum. The input overload points (I used visible clipping because of the hum level) were 6.3mV at 20Hz, 53mV at 1kHz, and 255mV at 20kHz.

PAiA 9802K

The PAiA preamp maintains normal output polarity. Input impedance was 46k7, and output impedance was a low 48Ω, all at 1kHz. The output noise was 0.14mV left (–81dB) and 0.28mV right (–75dB). The DC offset voltage was 130mV, probably reflecting the amplified input offset voltage of the 5532 op amp (4mV) in this direct-coupled design.

Gain at 1kHz, 10mV input was 43.9dB. The RIAA accuracy has an overall upward slope, as shown in *Fig. 8A*. Crosstalk at 10kHz measured –43dB in both directions.

The THD+N versus frequency (2mV/cm/s input level) is shown in *Fig. 9A*. Initially, the THD at 1kHz (in the higher right channel) was 0.13%. When I connected the inverse RIAA network chassis to the preamp ground post, it dropped to 0.008% on battery power and 0.012% with the AC adapter.

If I left the AC adapter plugged into the preamp with the transformer block unplugged from the AC outlet, and operated on battery power, the THD was 0.016%. One side of the 9V AC is connected to power-supply ground, so the lead length from the plug-in transformer to the preamp may be picking up noise. The bottom line is to unplug the AC adapter from the preamp if you use battery power. This would probably not be a problem with the PAiA regulated supply, which sends DC to the preamp.

The distortion residual showed the third harmonic with some very low-level noise. The THD+N versus line output level at 1kHz is shown in *Fig. 10A*. I ran this with the AC adapter, and again with 9V batteries. Both curves are shown in the figure.

The input overload at 1% THD clipping on battery power was too low for a 10mV cartridge: 5.7mV at 20Hz, 34.1mV at 1kHz, and 260mV at 20kHz. This is because the high gain of this preamp causes output overload, despite the highest power-supply voltage of all the tested units. When I connected the AC adapter (±15V DC supply), overload margin increased to an adequate 10mV at 20Hz, 58mV at 1kHz, and 440mV at 20kHz. This high gain, in conjunction with a high output cartridge, could overload some computer sound cards.

Since the PAiA preamp is a kit, you can easily make some improvements. As you can see from *Fig. 8A*, the PAiA preamp error curve is straight, but excessively bright. I recalculated the EQ com-

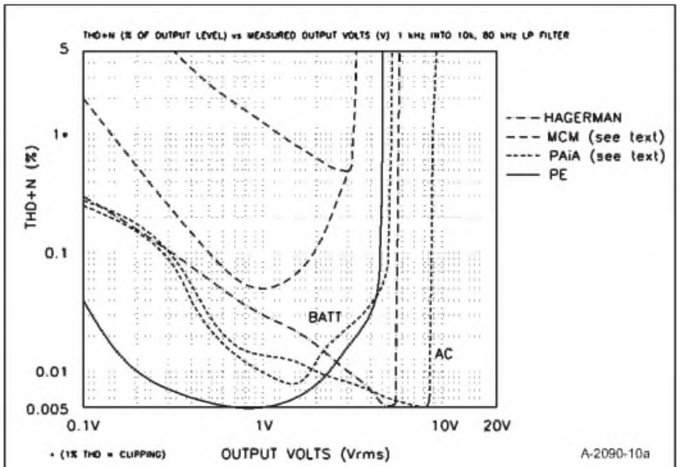


FIGURE 10A: THD+N versus output voltage at 1kHz (Hagerman, MCM, PAiA, PE).

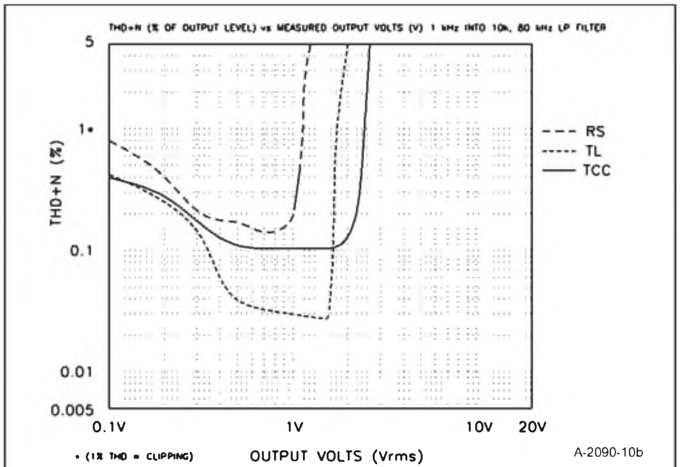


FIGURE 10B: THD+N versus output voltage at 1kHz (RS, TL, TCC).

ponents, using the Lipshitz formulas[5], leaving R5 and R9 at 470k. The nearest standard component values are listed below, and reduce the nominal error over the audio band to about ±0.25dB.

C4, C9	2n2
C5, C10	6n8
R4, R8	39k

These changes will also reduce the 1kHz gain by about –4dB, improving the input overload margin. You can further reduce the gain by increasing the value of R3 and R7. I would also replace the cartridge loading capacitor with a quality film or NP0 ceramic cap.

Here's one final possibility: The 5532 op amp has a maximum supply voltage rating of ±22V DC. You could use four 9V batteries to produce ±18V DC rails and achieve the highest headroom. If you install an 8-pin DIP socket in order to experiment with different dual op amps, be sure to observe the maximum supply voltage spec. The supply rails for many audio-grade op amps cannot exceed ±18V DC, and should not be operated above ±16V DC for reliability reasons.

PARTS EXPRESS PREAMP

The Parts Express #245-035 Rolls VP29 phono preamp maintains normal output polarity. Input impedance was 47k, and output impedance was 970Ω, all at 1kHz. The output noise was very low at 15μV, or –86dB.

When I made the DC offset voltage measurement, I found it wandered at a random ultra-low frequency about ±0.1V, almost like a record warp. At first I thought it might be the phantom ground to which the 47k input resistors are referenced, but the two inputs did not wobble together. Even when I loaded the outputs down to 1k, the wobble did not disappear. Grounding the inverse RIAA housing to the preamp binding post did not improve the situation. It was still there even after I connected the signal generator directly to the input, and may be due to high leakage in one of the aluminum coupling caps.

I don't think this wobble is of any sonic consequence. Most line preamps and power amps will roll off this less-than-1Hz component and not pass it on to the speakers. DC coupled amplifiers are another story, where this wobbly

DC offset will be multiplied by the amplifier gain.

Gain at 1kHz, 10mV input was 29.7dB. The RIAA accuracy was reasonably good, with an overall upward slope, as shown in *Fig. 8A.* Crosstalk at 10kHz measured –64dB in both directions.

The THD+N versus frequency (2mV/cm/s input level) is shown in *Fig. 9A.* I had some difficulty measuring distortion because of the randomly changing DC offset. The distortion test set has a fundamental notch filter with a feedforward circuit and automatic frequency tuning that tracks the input signal. When a wobble occurred, it caused the notch tuning to move slightly off frequency, decreasing the notch depth.

For higher-frequency THD measurements, I again used the 400Hz highpass filter in the test set. The lower frequencies required patience. I needed to wait for a lull in the DC offset wobble and grab the lowest THD reading I could see while the meter settled down. I think the THD performance of the PE preamp below 500Hz is probably better than that shown on the graph.

The distortion residual was mainly the third harmonic and some very low-level noise. The THD+N versus line output level at 1kHz is shown in *Fig. 10A.* Again, I took the data with the 400Hz HP filter. The input overload at 1% THD clipping was the best of the bunch: 19mV at 20Hz, 152mV at 1kHz, and 1.35V at 20kHz.

RADIO SHACK PREAMP

The RS 970-1018 phono preamp maintains normal output polarity. Input impedance was a low 26.5k, and output impedance was 450Ω, all at 1kHz.

The output DC offset voltage is negligible. The output noise, at 1mV (–54dB), is equal parts noise and 60Hz hum pickup. Grounding the preamp chassis to the inverse RIAA network housing caused the output noise to increase slightly.

Gain at 1kHz, 10mV input was 33.7dB. The RIAA accuracy was within the specified ±1dB within the audio band, but increased at lower frequencies, as shown in *Fig. 8B.* Crosstalk at 10kHz measured –58dB in both directions.

The THD+N versus frequency (2mV/cm/s input level) is shown in *Fig. 9B.* There is a slight dip at 60Hz, where the fundamental notch filter removed the 60Hz hum. The 1kHz distortion residual consists mainly of even-order harmonics, and low-level hum and noise. The distortion begins to rise rapidly above 16kHz, reaching 2.6% at 20kHz. While the response is flat, the circuit lacks sufficient HF headroom.

The THD+N versus line output level at 1kHz is shown in *Fig. 10B.* As the output voltage increases, the waveform becomes more and more triangular. The input overload points at 1% THD+N were too low: 2.4mV at 20Hz, 23.6mV at 1kHz, and 94mV at 20kHz.

TECH LINK

The Tech Link TPA2 phono preamp maintains normal output polarity. Input impedance was a low 24k, and output impedance was 890Ω, all at 1kHz. On 9V battery power, the output noise was low at 150μV, or –68dB. With the AC adapter, the output noise increased only slightly to 210μV, or –65dB. The DC offset voltage measured +2.7mV left and –0.1mV right. Crosstalk at 10kHz measured –64dB in both directions.

Gain at 1kHz, 10mV input was 31.6dB. The RIAA accuracy curve, as shown in *Fig. 8B,* had a LF peak at 75Hz, and then took off like a rocket above what should have been a pole at 2120Hz (75μs). Output clipping began at 11kHz. At 20kHz (not shown on the graph), the error is a huge +13.1dB, with 4.5 times more gain than is required. I don't know why the designers used an NAB equalization circuit for this preamp, but it produces a totally unacceptable equalization error when applied to a phono cartridge preamp circuit.

The THD+N versus frequency with 9V battery power (2mV/cm/s input level) is shown in *Fig. 9B.* The data was essentially the same with the AC adapter powering the preamp. I always select the higher channel for THD measurements, to show worst-case data. Usually the two channels are fairly close, but here the 1kHz measurements were 0.045% right and 0.13% left. That's quite a difference for a dual monolithic IC.

When I took the THD+N reading at 63Hz, I could see another hum pickup problem. The THD+N was 0.25% at 1kHz, but only 0.013% at 60Hz, where the test-set notch filter rejected the power-line hum. However, in this case, when I grounded the inverse RIAA network chassis to an unused phono jack shell on the preamp, the problem mostly disappeared, and I didn't need the 400Hz HP filter. In addition to fixing the EQ circuit, this preamp needs a dedicated ground connection point for the turntable ground lead.

The 1kHz distortion residual consisted mainly of low-level noise. Above 11kHz, the output began to clip on the lower half-cycles, and was in hard clipping by 16kHz. This unit will have major problems with a 10mV phono cartridge, and will be screechy and distorted at high frequencies even with lower sensitivity cartridges.

The THD+N versus line output level at 1kHz is shown in *Fig. 10B.* The headroom was unchanged with the AC adapter, even though it provided more voltage (9.9V DC versus 9.1 for a fresh battery).

The input overload at 1% THD below 2120Hz was marginal: 4.7mV at 20Hz and 50mV at 1kHz. The input overload was only 58mV at 20kHz.

The Tracertech website has detailed specifications for this preamp, although none came with the unit itself (*Table 1*). The preamp spec for THD is 0.004%, whereas the data sheet for the KIA6225 IC lists it as 0.04% typical and 0.25% maximum. The –65dB spec for preamp channel separation is the spec for the chip alone. Once you add peripheral circuits and interconnections, crosstalk will naturally increase. Compare the specs in *Table 1* with the measured data in *Table 2.*

TCC PREAMP

The TC-400 preamp maintains normal output polarity. According to the box, the input impedance is specified as a low 20k, but I measured 152k at 1kHz. Unlike the Radio Shack preamp, there

PREAMP SALE

Six of the preamps reviewed here are for sale through Old Colony Sound Lab (PO Box 876, Peterborough, NH 03458, 603-924-9464, Fax 603-924-9467, www.audioXpress.com, e-mail custserv@audioXpress.com), first come, first served, at the following prices, plus postage:

MCM Model #40-630	$10
PAiA 9802K	$20
Parts Express Rolls VP29 (with AC adapter)	$70
Radio Shack 970-1018	$21
Tech Link TPA2	$50
TCC TC-400 (with 12V DC adapter)	$21

is no discrete resistor at the input jack (my DMM read over 1M). I simulated the circuit with SPICE and got the same high input impedance. Output impedance was 995Ω, again at 1kHz.

The output DC offset voltage was only 3mV. The 1.4mV output noise (–60dB) is equal parts 120Hz hum and noise. Gain at 1kHz, 10mV input was 42.6dB. The RIAA accuracy, with a slope opposite all the other units, pretty much agreed with the Audio Precision graph on the box, as shown in *Fig. 8B.* Crosstalk at 10kHz measured –62dB in both directions.

The THD+N versus frequency (2mV/cm/s input level) is shown in *Fig. 9B.* The 1kHz distortion residual consists of low-level 120Hz hum and noise. The distortion below 100Hz was very high, increasing to 17% at 20Hz. Grounding the inverse RIAA network chassis to the preamp chassis made only a slight improvement.

The THD+N versus line output level at 1kHz is shown in *Fig. 10B.* As the output voltage increases, the waveform becomes more and more triangular. The input overload points at 1% THD+N were too low: 3.3mV at 20Hz, 11.5mV at 1kHz, and 63mV at 20kHz.

CONCLUSIONS

The discrete transistor circuit measurements were generally inferior to those of the op-amp units (the NAD 304 uses a discrete differential input stage). I would have to rate the Tech Link as unacceptable because of its NAB EQ curve. The two kit units offer great value.

The Hagerman Bugle preamp uses very high quality op amps, and passive EQ. RIAA circuits with passive EQ may clip a bit at the high frequencies because of the high gain required from the input stage, coupled with the rising cartridge output with frequency. By dis-

REFERENCES

1. "NAD PP-1 Phono Preamplifier," G. Galo, *Audio Electronics* 3/00, pp. 38–41.
2. "QED DS-1 Discsaver," M. Fremer, *Stereophile* 10/00, p. 52.
3. "Budget Wave Analyzer, Part 1," C. Hansen, *Audio Electronics* 3/99, p. 14, Fig. 7, without the three IC regulators.
4. The fourth RIAA time constant of 7950µs is defined in IEC Publication #98, Amendment 4, dated September 1976.
5. Letter, "PAT-5 Inversion and Equalization," S. Lipshitz, *TAA* 3/78, p. 48.

tributing the gain over three stages, the Bugle avoids this problem. The PAiA unit is also nice for the audio hobbyist who can easily modify it to improve its basic EQ and gain performance.

The Parts Express preamp measures the best "out of the box" unit, and it appears to have been well designed from the outset for its intended purpose. However, its gain is a bit on the low side. A little tweaking of the EQ parts values would make its RIAA error curve very flat.

Listening Critique

By John and Sandra Schubel

We listened to these seven budget-priced phono preamps, and compared them for sound quality, audible distortion, and background noise.

Our first observation is that the units from Hagerman, Tech Link, and Radio Shack are designed to operate off a 9V battery only. Our preference is that the preamplifiers be capable of AC operation, so that you can turn them on and off with the control amplifier.

A second observation is that two of the units—the Hagerman and the PAiA—are not enclosed. The Hagerman unit is neatly constructed and is available completely assembled, but its printed circuit board is exposed on the bottom, requiring care in its use to avoid shorting out the circuits. The PAiA circuit board is provided as a kit, so you must install it in a case such as a GEM box and connect it to Line Out connectors and AC filter prior to use.

The remaining units, the MCM Model #40-630, Parts Express Rolls VP29, Tech Link TPA2, and TCC TC-400, are all enclosed in boxes of various shapes and sizes. The Tech Link TPA2 has an added feature of an "AUX/PHONO" switch. This could be convenient if it would otherwise be necessary to disconnect some other source from your AUX input in order to connect the phono preamp.

LISTENING TEST

We used a Sony PS-LS520 turntable equipped with a Pickering TL-2S cartridge for the listening tests. We used the control amplifier and power amplifier sections of a Sony STR-DA555-ES, and the Sony's internal phono preamp as a reference. The speakers were a pair of NHT Model 1.3 speakers and NHT Model SW2 subwoofers.

We did the listening tests one audio selection at a time, because of the large number of preamplifier units to be reviewed. Although we used Sony's phono preamplifier as the reference, over time we were so pleased with the Hagerman that it became the benchmark for the others. We used a realistic sound pressure meter to assure that levels were consistent across all preamplifiers.

We did an initial screening of the preamplifiers using a direct-to-disk recording, *Lincoln Mayorga & Distinguished Colleagues Volume III*, Sheffield Labs SL5/SL6. Two cuts on this record—"America" from *West Side Story* and "That Certain Feeling"—are clean and offer wide frequency response and percussive sounds.

We performed subsequent tests using these recordings:

"The Sea and Sinbad's Ship" from *Scheherazade* (Rimsky-Korsakov), London SPC 21005, Leopold Stokowski and the London Symphony Orchestra.

"Surely He Has Borne Our Griefs" and "All We Like Sheep Have Gone Astray" from *Messiah* (Handel), L'oiseau-Lyre D189D3, Christopher Hogwood and the Academy of Ancient Music and the Choir of Christ Church Cathedral, Oxford.

The Moldau (Smetana), RCA Red Seal LSC-2471, Leopold Stokowski and the RCA Victor Symphony.

HAGERMAN BUGLE

The Hagerman unit had no audible noise or hum at both normal and extended volume levels. The sound with "America" was not as bright as with the Sony, although cymbals and triangles were crisp. Instrument detail was excellent, and the listener was brought close to the soundstage. Particularly impressive was the detail given to the saxophone on the second track, "That Certain Feeling." The impression is that there is more midrange from the Hagerman than from the Sony.

Scheherazade sounded much cleaner on the Hagerman than on the Sony preamp. The violins were exceptionally sweet and clear, as were the harp, cello, and woodwinds. Complex musical passages were handled cleanly. You had the impression of being in the middle of the concert hall.

Handel was very clearly reproduced, with each section and solo voice clearly articulated. The violins and harpsichord were also clearly reproduced. The effect was as if you were there, singing in the choir loft.

The Moldau was slightly dull on this preamp. The triangles and piccolos were clear, the bass was full, and the violins were clear except in the most complex passages. The sound was more like the back of the concert hall. You had to strain to pick out the clarinet in the opening measures of the piece. The tympani were well reproduced, as were the trumpets, yet the sound remained distant. It was difficult to place instruments on the soundstage.

PAiA (BATTERY)

The PAiA unit had no audible noise or hum at both normal and extended volume levels. The sound with "America" was extremely bright, and sometimes harsh with horns. The guitars seemed out of balance with the other instruments. The first impression was that this preamplifier would tire the listener easily.

Scheherazade sounded brighter on the PAiA than on the Hagerman. The violins and harpsichord stood out from the other instruments. The sound of the harpsichord was very pleasant, but the violins became strident as the score modulated higher and higher. The cello sounded somewhat tinny. The sound became muddied during loud complex passages.

Handel was unpleasantly bright on this preamp. The choir sections were not well defined. The violin and harpsichord were very bright. The treble choir was reproduced with great clarity, but it was "in your face." Although the sound was bright, the listener did not get the sense of being close to the performers.

The flutes and clarinets, and the plucking of the violin at the beginning of *The Moldau*, made you feel that you were at the front of the audience, as did the striking of the cymbals. The bass was full and rich, and the violins stood out with clarity. It was easy to place the instruments on the soundstage, and there was no muddiness even in loud and complex passages.

Sometimes the sound became too bright, particularly with loud brass passages. You believed that you were at the front of the concert hall. John liked listening to this piece with this preamplifier, although perhaps he would have reduced the volume below that which we had established as a reference for this test.

PAiA (AC ADAPTER)

The PAiA unit had no audible noise or hum with the AC adapter at either normal or extended volume levels. The sound was indistinguishable from that experienced under battery power. The sound remained very bright, somewhat masking the percussion. Since there was no discernible difference in sound with the AC adapter, we decided to do the balance of testing with battery power.

MCM

The MCM preamp is powered by 120V AC, with the power supply internal to the preamplifier case. This preamplifier had significant hum. The power plug is polarized, so no attempt was made to reverse the polarity. The turntable was grounded at the Sony control amplifier, as there is no provision on the preamp for grounding the turntable.

The sound on "America" was less bright than the PAiA, and slightly brighter than the reference Sony. This preamp seemed to bring out the attack of the drums, and muddied the harpsichord. The raspiness of the saxophone on "That Certain Feeling" was very muddy.

The first thing we noticed when playing *Scheherazade* was that as soon as we set the needle down in the groove the hum level grabbed our attention. The violins screeched as the score modulated upward. The cello sounded raspy and the clarinets had no personality.

In general, it was difficult to clearly identify the various instruments. The sound was muddy on complex passages. One interesting phenomenon when using this preamp was that touching the controls on the amplifier or turntable sometimes produced a "pop," and there was always a loud "pop" when the turntable set the needle down on the record.

Handel was pleasantly reproduced, if you ignored the constant hum and the loud "pop" as the needle set down on the record. The sections were clearly defined and well separated. Soloists stood out well from the chorus. This preamplifier produced a surprisingly clean playback of this selection . . . if it were not for that hum.

The listening test with *The Moldau* was inconclusive as low-level passages were masked by the hum of the preamplifier. John set the flute entrance to 60dB, which yielded a listening level of approximately 80dB for all but the loudest passages. When he attempted to set this level for the MCM, the hum was as loud as the flutes, invalidating his setting. Giving it our best shot at level setting, the amplifier otherwise sounded fairly clean, but placed the listener at the back of the hall.

ROLLS VP29

The Rolls preamp is powered by a 12V DC transformer supplied by the manufacturer. This preamp had no audible hum or buzz at normal or elevated listening levels. When the volume was advanced on the Sony's control amplifier, it consistently tripped the unit's protection circuitry, shutting down the amplifier. This happened both with the turntable grounded at the receiver and at the preamplifier.

This preamplifier had a distant sound, and lacked clarity in either the midrange or the high end. We could not attribute it to lack of high-frequency response, although the high end was subdued. In general, the performance of "America" and "That Certain Feeling" sounded distant and unpleasant. Instruments lacked clarity.

We reconnected the preamplifier the next day, intent on setting levels with the realistic sound-level meter, and then playing *Scheherazade*. Before we could adjust the level, we heard a pulsing buzz from the amplifier, consisting of two buzz pulses one second apart, and then two seconds of silence before the two buzz pulses repeated. We listened to this for about 15 seconds and shut down the preamplifier. We tried no further testing on this unit in deference to the Sony and the speakers.

We contacted the distributor of this preamplifier, and they provided a second sample, which also consistently tripped the Sony's protection circuitry when we advanced the volume. This second sample also produced no buzz or hum until just before the Sony's protection circuit activated. As the volume was advanced, we suddenly heard a buzz, followed almost immediately by the amplifier shutting down and the word "Protection" appearing across the amplifier's display.

We repeated the listening test with "America" and "That Certain Feeling" and again thought the preamplifier had a distant sound. What was really noticeable was the difference in gain between, for example, the Hagerman preamplifier and the VP29, with the VP29 requiring approximately 11dB more gain from the Sony to produce the same listening level. John took the realistic sound-level meter and noted that we preferred a listening level of, on average, 82dB. To maintain this level with the VP29, we needed to crank up the level to near where the amplifier protection circuit activated.

Curious about how much we could increase the listening level, we risked fate just one more time and cranked up the volume. The protection circuitry activated at approximately 8dB above the reference volume level. We backed the volume down slightly, turned the amplifier back on, and verified that the average listening level was now around 90dB. We also noted that when we stopped the record, we could now hear a pulsing buzz similar to that which we experienced with the first sample.

We backed the volume down to the reference level and listened to *Scheherazade*. The violins sounded brittle, lacking the warmth experienced when listening with the Hagerman preamplifier. When the harp entered, we sensed that the recording of harp and violin was made in a room with metal walls. The full orchestra did not fare better, sounding distant and tinny. Loud passages sounded shrill and distorted.

The Moldau also gave the impression that the orchestra was far away. It was difficult to place instruments on the soundstage, and we needed to strain to pick out individual instruments, which at times sounded shrill, yet high-frequency percussive sounds such as triangles did not stand out.

Handel also sounded distant on this preamplifier. The choir fared well, as did soloists. It just sounded as though we were at the back of the church. The instruments were muted to the point where the choir dominated the sound. We knew there was a harpsichord in there, but we had to hunt for it.

TECH LINK

The Tech Link preamplifier sounded very clean and slightly bright when playing "America." It produced no audible hum at normal listening levels, or at extended listening levels. It gave the listener the impression of being very close to the performers, but the sound was sufficiently clean that the brightness was not objectionable.

Instruments were well defined and separated. The harpsichord was very clean, and the saxophone sound in "That Certain Feeling" was similar to that produced by the Hagerman. Trumpets were very cleanly presented. Cymbals were very crisply presented, bordering on harsh.

Scheherazade on the Tech Link was very bright, and the violin and harpsichord stood out from the other instruments. The violins overpowered the other instruments on loud and complex passages. Most instruments in the orchestra—when they had the chance during quiet passages—were clearly defined. We thought we were sitting in the violin section. The sound was very clean, but the overemphasis of brass and violins grew tiring.

Handel was very bright, but all sections were cleanly reproduced. Violins and harpsichord stood out clearly. You had the feeling that you were in the front row. Sections were well defined and separated.

The choir tended to overpower the instruments. Sandra thought that the voices tended to morph into instruments and be flat, not dimensional. John normally loves the front row, but for this performance it was a little too close. Reducing the listening level would have made this as pleasant to listen to as the Hagerman.

The Moldau was also very bright. Sandra found it to be "edgy." This was OK with John on the opening passages, as it made the violins, triangle, and wind instruments stand out with clarity.

Later in the piece, it caused the violins and triangle to become overbearing, out of balance with the remainder of the orchestra. The sound never became cacophonous, just too bright. It was difficult to place the listener in the hall.

RADIO SHACK

The Radio Shack unit produced no audible hum or hiss at normal or elevated listening levels. Playing "America" on this unit produced a more distant sound than the Sony, Tech Link, or Hagerman, and seemed to lack definition. The kick drum did not thump with authority, and the cymbals and triangle did not seem as bright. John rechecked his levels to be sure that this phenomenon was not caused by reduced listening levels, and found them to be correct.

Scheherazade was very listenable on the Radio Shack preamp. When individual instruments played solos, they stood out well. Violins and harpsichord were cleanly reproduced, as was the cello. Woodwinds did not stand out with their distinctive personalities. The instruments were not well defined during loud passages, but the effect was more of being further back in the hall, as opposed to being muddy.

Listening to Handel, the violins and harpsichord were well reproduced, but the organ sounded tinny. The choir sections and soloists were cleanly reproduced. We sensed the reverberations in the hall. As such, individual sections did not stand out, although the sound never became muddy. All and all, it was what you might expect if you were seated in the middle of church.

Our initial impression of *The Moldau* was that the performance was quieter than with other preamps, even though the level was set using the sound-level meter. The plucking of the violins during the opening measures was not crisp, and John believed it was difficult to identify whether the accompanying instruments were flutes or clarinets. Sandra, on the other hand, particularly liked the sound of the flautist's breath across the mouthpiece.

Once the full orchestra came in, the bass was solid and full, and the general balance of the orchestra was excellent. The sound seemed brighter than with the Hagerman but less so than with the PAiA. Still, instrument detail was missing. The oboe, for instance, did not have its characteristic overtones.

TCC TC-400

The TCC unit produced no audible hum or hiss at normal listening levels. We could hear a slight hum with the Sony turned to full volume. The turntable was grounded to the Sony, because there was no provision on the preamplifier for local grounding. This unit produced a sound akin to the Hagerman when playing "America." The sound was very clean, and the kick drum was struck with more authority than on other preamplifiers.

The sound was full across the sound spectrum, and instruments were well balanced. The saxophone seemed slightly subdued in "That Certain Feeling." The listener seemed to be placed near the performers, although not as close as with the Tech Link. Out of curiosity, I compared the Hagerman directly against the TCC, and confirmed that the TCC appeared to have more emphasis at the very low bass.

Since this phenomenon had not shown up on other preamps, John did a little experiment with an old Theatre Organ recording. The test vehicle was a recording of "It's Almost Like Being in Love" (Lerner and Loewe) on the Paramount Theatre (New York) organ immediately before the organ was packed up and the building torn down (Command RS 881 SD, *Showtime*, Ashley Miller, organist). The Sony and Hagerman preamps sounded very similar on this recording, but the TCC was significantly heavier in the low bass.

Scheherazade had a warmer sound than with the Hagerman. Violins did not sound as sweet, and the cello sounded tinny. Individual instruments did not stand out with clarity in loud, complex passages. The sound placed the listener farther back in the audience. On occasion the sound seemed slightly harsh, and the bass clarinet lost its tone.

Handel seemed to lack clarity. The sections of the choir seemed to blend together and lose their individual personalities. The cello was obtrusively loud, and the harpsichord sometimes became lost behind the chorus. We found ourselves straining to hear the sections with the same clarity that we heard from some of the other preamplifiers, particularly the Hagerman. The reverberation of the hall was muddy. The performance was unremarkable when played on this preamp.

Listening to *The Moldau*, this preamp initially sounded slightly bright, as the triangles and violins seemed to stand out. We could clearly identify the flutes and clarinet. As for the soundstage, we thought we were in the middle of the hall. Once the bass began to play, the spell was broken. The bass was too loud and heavy, which dampened our enthusiasm for an otherwise enjoyable listening experience.

CONCLUSION

The clear winner from this assortment was the Hagerman preamplifier. Its performance was consistent across all recordings. John's second choice in this group was the Tech Link, although he would depend on the control amplifier to tone down the treble. Sandra would go with the Radio Shack unit as her second choice, because she found the Tech Link too bright.

The next two on our list were about on par, each with its own personality. The PAiA is too bright in the treble range, and the TCC accentuates the low bass.

The remaining two units—the MCM and Parts Express—were unacceptable. The hum on the MCM would make it unusable, especially for copying your record collection to CD. The Parts Express units consistently tripped the protection circuitry of the Sony amplifier.

Now the hard question is for which one from this group of seven would we spend our money? If your use is occasional, the Radio Shack unit is hard to beat as a compromise of performance, price, and availability. If you play your records regularly, the nicely packaged and AC-powered Tech Link is a good choice. If your objective is to copy your prized record collection to CD, and neither the open circuit board nor battery operation is an issue, the Hagerman Bugle is the one. ❖

Manufacturer's Response:

Thank you for the informative and timely article on budget phono stages. Just to make sure there is no confusion on pricing, the Bugle "half-kit" is only $25. A completed unit runs about $50 in total parts.

Regarding RIAA response, that rise in Bugle response above 10kHz is intentional. It is to compensate for the 50kHz turnover which most products ignore. In fact, I believe the particular inverse filter used in testing is incomplete by omission. This is all explained in my Audio Electronics *3/99 article "On Reference RIAA Networks" (filter available at www.hagtech.com). So as I test the Bugle, RIAA response is ruler-flat across the audio band. I would not hesitate to claim it as the most accurate of any production phono stage, independent of price. Square-wave performance is phenomenal. The 1% resistors and 2% polypropylene capacitors guarantee consistency.*

Similarly, I expect the small bumps in the distortion curves are the result of difficult test conditions. The dip at 60Hz is probably due to the analyzer's notch and the peak at 120Hz from a stray instrument hum. The Bugle itself is inherently immune from hum as it is powered from batteries. Nevertheless, strong AC fields in close proximity can couple into the unshielded circuitry.

I am honored by the comments "over time we were so pleased with the Hagerman that it became the benchmark for the others" and "the clear winner from this assortment was the Hagerman preamplifier." I designed the Bugle as a statement piece not to compete with other budget amplifiers, but against far more expensive units. Many happy customers confirm that it achieves true audiophile quality. ❖

Jim Hagerman
Hagerman Technology LLC

Testing the SEAS Froy Mk3

By Joseph DAppolito and Dennis Colin

According to the SEAS website (www.seas.no), the Froy (Mk3) kit is an improved version of the original Froy kit. It uses SEAS' latest drivers from the EXCEL line. The kit employs two W15CY001 15cm magnesium cone woofers with one EXCEL T25CF002 Millennium tweeter in my MTM configuration. Ed Dell built the two Froy (Mk3) loudspeakers tested here. (March '03, p. 60)

I ran a series of impedance, frequency response, polar response, and distortion tests on the Froy (Mk3). *Figure 1* is a plot of system impedance magnitude. At low frequencies the plot displays the double-peaked curve typical of vented systems. The impedance minimum of 3.74Ω at 54.2Hz indicates the vented-box tuning frequency (f_B). There is a second impedance minimum in the low-frequency range of 3.2Ω at 210Hz. Impedance phase angles range from +34° to −43°. I would rate this a 4Ω speaker.

FREQUENCY RESPONSE

Figure 2 shows the full-range frequency response of one Froy (Mk3). This response is obtained as a combination of the far-field quasi-anechoic response and properly summed near-field woofer and near-field port responses. I placed the microphone along the tweeter axis at a distance of 1.25m to produce the far-field response. The near- and far-field responses were then spliced together at 200Hz to produce the full-range response[1]. The response data is 0.1 octave smoothed.

In the octave between 500Hz and 1kHz, sensitivity averages 87.6dB SPL/2.83V/1m. This is only 0.4dB less than the 88dB SEAS claims and well within acoustic measurement accuracy. Relative to the 87.6dB level, response varies by +2.1dB and −1.2dB over the range of 200Hz to 20kHz. The +2.1dB point occurs in the tweeter's response range at about 4.5kHz. The −3dB low-frequency point is 72Hz.

The rather poor low-frequency extension is surprising. A quick analysis in LEAP shows these woofers capable of an f_3 of 55Hz in a QB3 alignment with a 12 ltr box tuned to 43Hz. Contrast this with the measured f_B of 54.2Hz. (The second sample measures even higher at 56Hz.) SEAS claims a low-frequency response to 40Hz, but no limits are given on this figure. SEAS' own plot shows the Froy down about 14dB at 40Hz when measured in their anechoic chamber.

The Froy (Mk3) impedance curve given on the SEAS website shows an impedance minimum of about 47Hz. A quick glance at the Froy plans shows that Editor Dell has reproduced the enclosures accurately, which leaves the discrepancies in f_B unexplained.

Figure 3 plots system and individual driver responses between 200Hz and 20kHz. This plot shows the crossover frequency for this sample to occur at 2061Hz, somewhat below the 2200Hz claimed in the SEAS literature. Notice the woofer pair response peak at 8.24kHz. SEAS claims the woofer peak is suppressed in the crossover with a series LC network in parallel with the woofers. However, examination of the crossover schematic on the website shows a Zobel across the woofer terminals, which negates the effect of the shunt. More on this later.

WOOFER/TWEETER TIMING

The Froy (Mk3) step response is plotted in *Fig. 4*, which shows two separate arrivals of acoustic energy. The initial sharper positive spike is the tweeter arrival. It is followed by the woofer arrival, beginning about 0.2ms later. Although not shown, a detailed examination of the excess group delay plot[1] shows the woofer pair to be 200μs (0.2ms) behind the tweeter. Although all drivers are connected with positive polarity, the system is not time-coherent.

CUMULATIVE SPECTRAL DECAY

The Froy (Mk3) cumulative spectral decay (CSD) response is presented in *Fig. 5A*. This waterfall plot shows the frequency content of the system response following a sharp impulsive input at time zero. On the CSD plot, frequency increases from left to right and time moves forward from the rear. Each slice represents a 0.11ms increment of time. The total vertical scale covers a 30dB dynamic range.

Ideally the response should decay to zero instantaneously. Inertia and stored energy that take a finite amount of time to die away, however, characterize real loudspeakers. A prominent ridge parallel to the time axis indicates the presence of a strong system resonance.

The first time slice in *Fig. 5A* (0.00ms) represents the system frequency response. The major decay response above 3kHz falls 30dB in 0.8ms. However, there is a fair amount of "hash" beyond this point. In particular, there is a ridge at 8.2kHz extending out beyond 3ms. This result is at first surprising since the EXCEL Millennium tweeter does not display this poor decay response in the THOR transmission line[2].

Figure 5B gives the source of this hash. This plot of woofer pair response without smoothing shows a strong response peak at 8.2kHz followed by successively smaller peaks at 10.8, 13, 15.7, and 18.5kHz. These are the higher frequency breakup modes of the EXCEL woofer cones. The major peak is only 13dB below the full system response. *Figure 5C*, a CSD of the woofer pair, shows these modes clearly.

Returning to *Fig. 3*, you see that the woofer pair response falls off only 9dB in the octave above crossover and the primary woofer breakup peak is not suppressed. Contrast this with Fig. 14 in reference 2, where response falls off 15dB in the first octave above crossover and the primary woofer peak is fully suppressed. Careful listening will determine what, if any, effect the untamed woofer modes have on sound quality.

The low-frequency decay is rather rich in frequency content and extends out to about 4ms. This is fairly typical of vented loudspeakers.

HORIZONTAL POLAR RESPONSE

Horizontal polar response is examined in *Figs. 6* and *7*. *Figure 6* is a waterfall plot of horizontal polar response in 10° increments from 60° right (+60°) to 60° left (–60°) when facing the speaker. All off-axis plots are referenced to the on-axis response, which appears as a straight line at 0.00°. For this reason, the plotted curves show the change in response as you move off-axis. For good stereo imaging the off-axis curves should be smooth replicas of the on-axis response with the possible exception of some tweeter rolloff at higher frequencies and larger off-axis angles.

Within ±30° the off-axis curves are indeed fairly smooth replicas of the on-axis response. The –3dB coverage at 15kHz is ±25°, which is typical of 25mm dome tweeters. The 60° curve shows the transition from the woofer pair to the tweeter.

At 1.6kHz, response is down 6.7dB relative to the on-axis response. This is due to the woofer pair directivity at that frequency. At 2.6kHz, however, the 60° off-axis response is down only 2.4dB, because the system output has transitioned from the woofer pair to the tweeter. The response at large off-axis angles is typical of two-way systems.

The average response over a 60° horizontal window (±30°) in the forward direction is a good approximation of the way a speaker will sound in a typical listening environment (*Fig. 7*). This response is within 1dB of the on-axis response out to 10kHz and is only 2dB down at 15kHz. This is excellent horizontal performance and suggests good direct field coverage in the primary listening area with little change in spectral balance with changing position. Image stability should be very good.

VERTICAL POLAR RESPONSE

Figure 8 is the waterfall plot of vertical polar response. Responses are shown in 5° increments from 25° below (–25°) the tweeter axis to 25° above it. Off-axis responses out to ±10° track the on-axis response with little error.

As angles approach 20° and more, deep symmetric notches develop just below 2kHz. This performance is typical of the MTM geometry, and is actually one of its major advantages. This vertical off-axis response greatly reduces floor and ceiling bounces that tend to confuse imaging.

Figure 9 plots the average vertical polar response over a ±10° window. This average tracks the on-axis response within 1dB out to 10kHz and is down only 1.6dB at 15kHz.

HARMONIC DISTORTION

I ran harmonic distortion tests at an average level of 90dB SPL. Ideally, harmonic distortion tests should be run in an anechoic environment. In practice, it is impor-

CRITIQUE

Reviewed by Dennis Colin

I judge the Froy speakers to be very good except for two factors: (1) thin-sounding bass and (2) a mild but "brassy" emphasis around 3–4kHz. The latter was not noticeable on all material. But on horns and strings, and sometimes on a well-recorded powerful voice, I heard some roughness or congestion, best describable as not so much frequency emphasis, but rather some harshness with confusion of detail in the upper midrange.

To put this into perspective, the overall sonic naturalness was about 80% of the way from the Adire speaker (which I highly criticized) to the THOR (which I thought second to none in tonal quality). On much material the Froys sounded very good. And on all material, the highest treble smoothness and general low-mid-high balance were excellent.

Regarding image focus, soundstage realism, and spatial reproduction, these were outstanding, perhaps as good as the THOR (or at least very close).

SPECIFIC IMPRESSIONS

Turtle Creek Chorale—Voice very good, not quite as natural as with THOR; bass sounded thin but not colored.
A Chorus Line—Somewhat thin, upper mids slightly "brassy," still good-sounding overall. Estimated bass extension to 60Hz.
Jacintha—Here, the voice midrange sounded natural, but highs somewhat sibilant (roughness on "S" sounds).
Carmen, Percussion Fantasia—Bells excellent, super image focus and stereo staging; perceived some emphasis around 3–4kHz (est.).
Carmen Ballet—Good, neutral tonality throughout.
Beethoven Pastoral—Good, but bass was too thin.
Chopin—Excellent piano rendition and presence.

COMMENTS ON MEASUREMENTS

Figure 2—The LF –3dB point of 72Hz explains the thin bass perfectly well. The mild peak at 4.5kHz might explain the emphasis heard (I had estimated 3–4kHz).

Figure 5A—Joe mentioned the ridge at 8.2kHz. This explains the sibilance on Jacintha's HF-rich voice. But I didn't notice this on other material. Rather, I think the pronounced long-decay ridge around 3kHz was what bothered me the most, more so than the mild 4.5kHz peak on the first-arrival frequency response (0ms top of waterfall, all Fig. 2). The former 3kHz "hashiness" of decay would be the most likely contributor, I think, for the upper-mid roughness or congestion I heard. Such a decay pattern would also likely obscure fine details.

Figure 5B—The woofers' 8.2kHz resonance, even though largely obscured on Fig. 2 by the crossover to tweeter, nevertheless was sometimes audible.

Figure 5C—Surprising here is the absence of the 3kHz ridge from the combined waterfall of Fig. 5A. If the long 3kHz decay isn't from the woofer, and most likely not from the excellent Millennium tweeter, where does it come from?

My guess is (by process of elimination) that the crossover is causing some ringing (undamped resonance). While not evident in the overall frequency response, Fig. 4 (step response) might provide a clue. Notice from 3.4ms on the time axis to 4.2ms a small but nearly periodic wiggle, with cycle peaks around 3.4, 3.75, and 4.1ms. With a cycle period around 0.35ms, my alleged ringing frequency would be about 2.9kHz, close enough to the approximate 3kHz prominent ridge on Fig. 5A.

Or perhaps the tweeter LF rolloff area around 3kHz (Fig. 3), where you see a mild but fast 2.5kHz peak-to-2.9kHz dip transition, represents (I further speculate) a lack of tweeter damping by, say, too high a driving impedance at this frequency range.

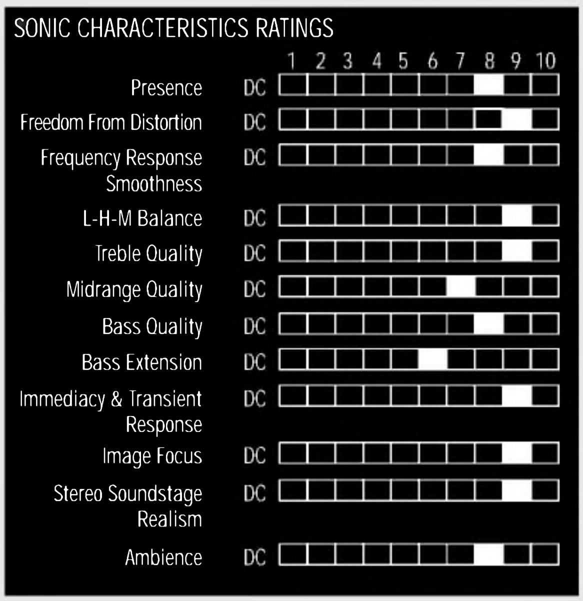
Figure 6—This smooth, well-behaved polar rolloff is consistent with the excellent imaging and sound-field reproduction.

Figure 7—Smooth overall. However, while the peak at 4kHz is small in dB value, you see a noticeable "corner-like" effect. In my experience, such slope-changing effects are audible.

RETROSPECT

I must reiterate that despite my having fun playing detective, the colorations noted were small on most material, and unnoticeable on some. The Froys are in most aspects excellent speakers. I believe that, considering the identical tweeter and similar woofers (to the THOR speaker), the Froy anomalies are crossover-related.

I think the moral of this story is: If you want the best possible crossover, have Joe design it. I hope he receives permission from SEAS to do just that here; if so, I look forward to auditioning the result. But even as it is, the Froy (Mk3) speakers are very good; the SEAS' drivers are probably among the world's best.



tant to minimize reflections at the microphone during these tests. Out-of-phase reflections can produce false readings by reducing the level of the fundamental while boosting the amplitude of a harmonic. In order to reduce the impact of reflections, I placed the microphone at 0.5m from the loudspeaker and gated response to largely eliminate later reflections.

Second and third harmonic dis-

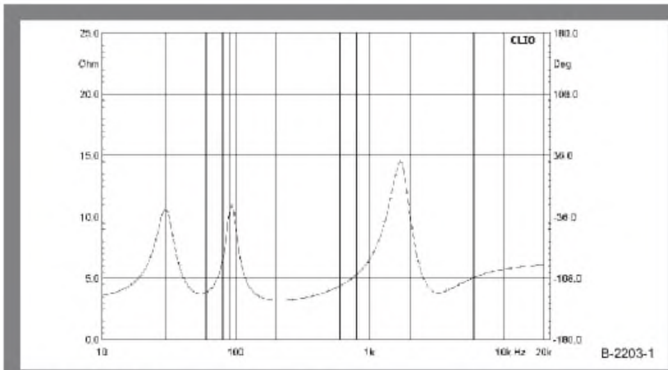


FIGURE 1: SEAS Froy (Mk3) impedance.

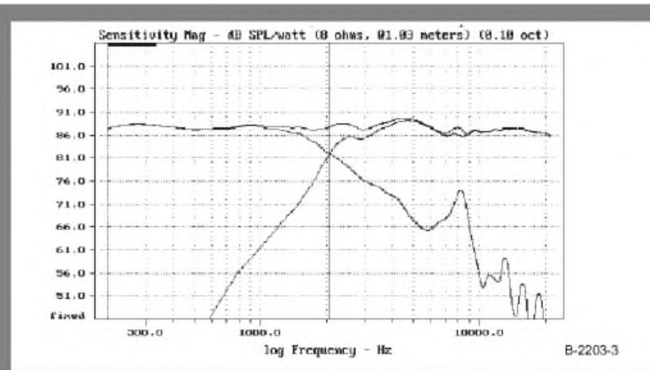


FIGURE 3: Froy system and driver responses.

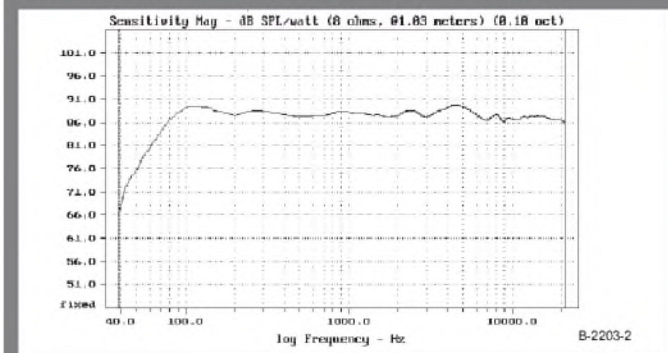


FIGURE 2: Froy (Mk3) full-range frequency response.

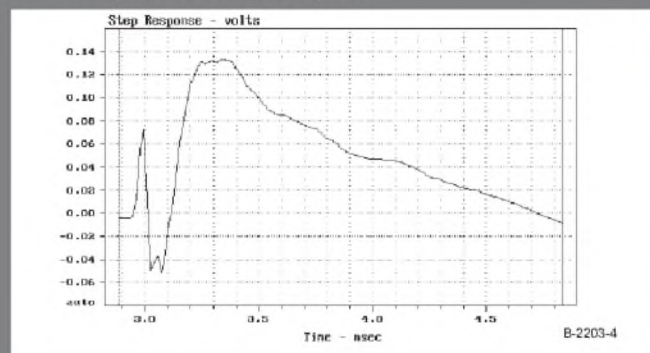


FIGURE 4: Froy (Mk3) step response.

tortions at 50Hz and 90dB SPL were 5.4% and 5.0%, respectively. 50Hz lies below f_B, and 90dB SPL places quite a demand on these small woofers. However, all harmonic distortion falls below 1% above 100Hz, which is a very good result.

INTERMODULATION DISTORTION

I measured intermodulation distortion next. In this test two frequencies are input to the speaker. Intermodulation distortion produces output frequencies that are not harmonically related to the

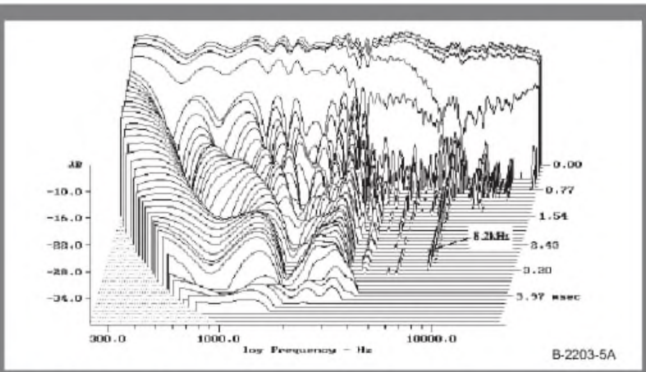


FIGURE 5A: Froy (Mk3) cumulative spectral decay.

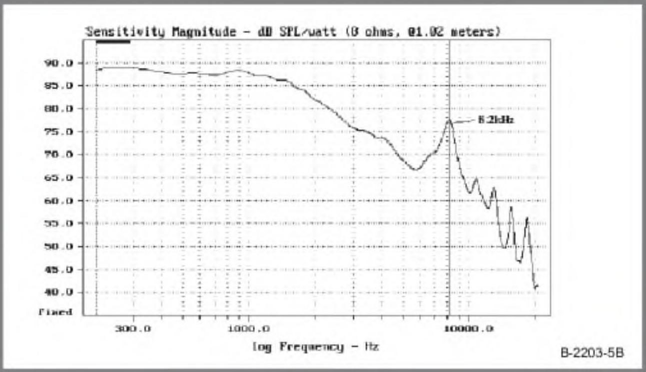


FIGURE 5B: Froy (Mk3) woofer pair response.

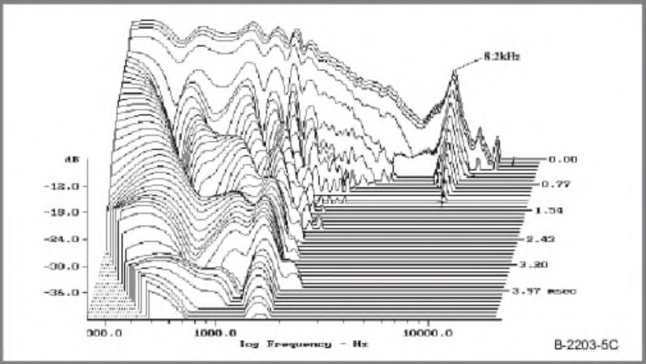


FIGURE 5C: Froy (Mk3) woofer pair CSD.

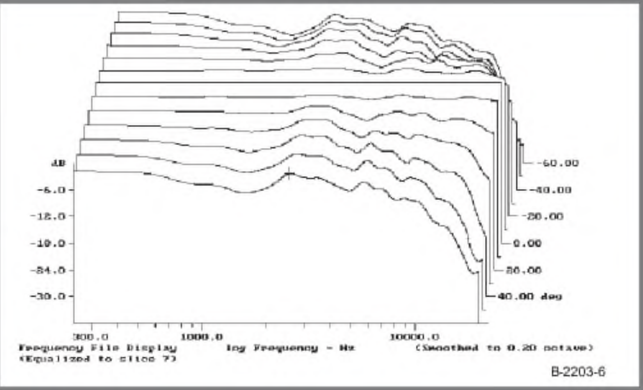


FIGURE 6: Froy (Mk3) horizontal polar response waterfall.

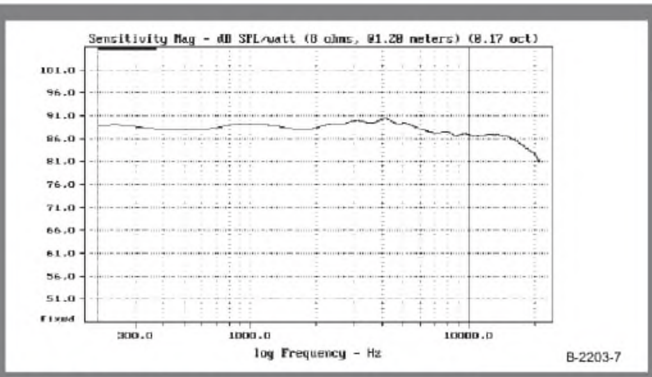


FIGURE 7: Froy horizontal response averaged over ±30°.

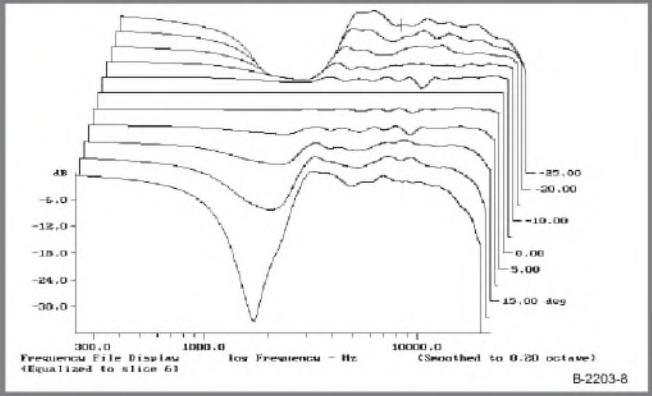


FIGURE 8: Froy vertical polar response waterfall.

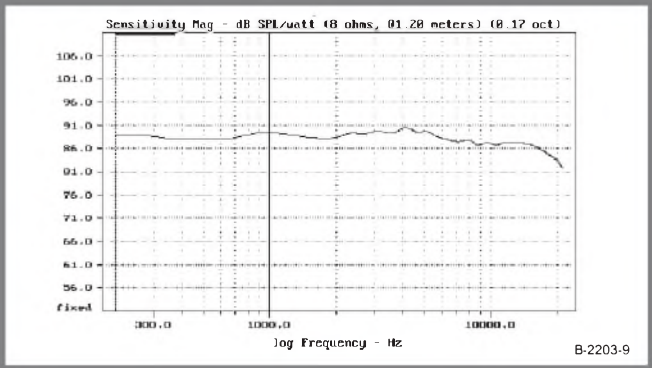


FIGURE 9: Froy (Mk3) average vertical response ±10°.

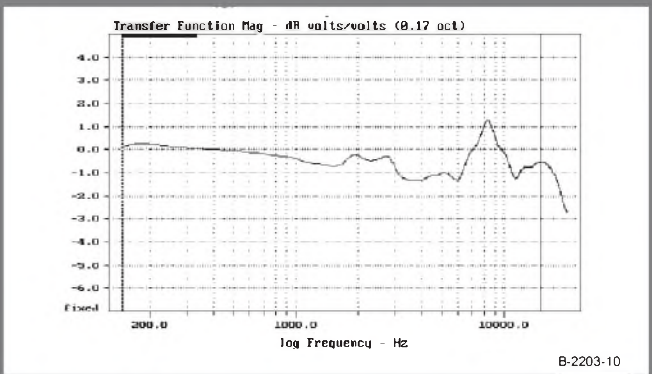


FIGURE 10: Froy (Mk3) sample response difference (#2-#1).

input. These frequencies are much more audible and annoying than harmonic distortion.

Let the symbols f_1 and f_2 represent the two frequencies used in the test. Then a second-order nonlinearity will produce intermods at frequencies of $f_1 \pm f_2$. A third-order nonlinearity generates intermods at $2f_1 \pm f_2$ and $f_1 \pm 2f_2$.

I examined woofer intermods first by inputting 400Hz and 550Hz signals at equal levels. These frequencies should appear predominantly in the woofer output. Total SPL with the two signals was adjusted to 90dB at 1m. Significant woofer IM products appeared at 950, 1350, and 1500Hz. However, the overall level was only 0.09%, an excellent result.

I measured tweeter intermods with a 9kHz and 10kHz input pair adjusted to produce an 87dB SPL at 1m. Because steady tones are used in the IM test, I thought it safer to use a lower power level to prevent possible tweeter damage. The major IM product occurred at 12kHz. However, total distortion was only 0.07%, a very good result for tweeters.

The last IM test examines cross-intermodulation distortion between the woofer and tweeter using frequencies of 900Hz and 10kHz. Ideally, the crossover should prevent high-frequency energy from entering the woofer and low-frequency energy from entering the tweeter. IMD products appeared at 8.2, 9.1, and 10.9kHz at an overall level of 0.05%. This is a very good result and indicates good inter driver isolation by the crossover.

SPEAKER MATCHING

All of the test results reported so far were obtained from a single sample. Now look at how well the two speakers match in frequency response (*Fig. 10*). The two Froy (Mk3) samples match quite well. The second system is within ±1.3dB of the first out to 15kHz. This bodes well for image stability.

CLOSING REMARKS

Many of the points raised in the test review may seem less than complimentary to the Froy (Mk3). I mentioned them only to highlight some of the subtleties that you can encounter in loudspeaker testing. On balance, the results reported here in the areas of frequency response, polar response, and distortion are very good. I suspect that auditioners will be very impressed with the Froys.

Manufacturer's Response:

Our sincerest thanks to Joe D'Appolito, Dennis Colin, and Ed Dell for their excellent in-depth review of the SEAS Froy Mk. III kit. We are generally in agreement regarding Joe's measurements and Dennis' listening evaluations, but there are a couple of points we would like to address.

Joe points to a discrepancy between the cabinet tuning frequency of the test samples compared with the Froys that were built and measured at SEAS. We, too, are puzzled by his results, as the f_B of his systems were indeed 7–10Hz higher than ours. About the only thing we can point to is the possibility that the port slot openings in Ed's cabinets are slightly smaller than specified. This is quite critical, because even a very small change in the height of the slot will have a significant impact on the area of the port.

We also agree with Joe that the W15 CY001's 8.2kHz resonance peak could have been better suppressed by the crossover's notch filter. Thankfully, this peak is at a rather high frequency, and is still sufficiently down in level to have minimal impact on the frequency response and distortion measurements of the system. Still, we will revisit the crossover design in the future to see whether this aspect can be further improved.

Regarding Dennis Colin's listening critique, he faults the Froy in two areas: bass extension and a mild emphasis around 3–4kHz. We believe the two are related. While the overall balance of the Froy is basically flat, the lack of deep bass response will cause the system's perceived balance to be shifted more towards the upper end of the spectrum; i.e., it will sound a little bit bright. Additionally, when comparing the horizontal dispersion characteristics of the THOR with those of the Froy, you will find that the Froy's smaller woofers are capable of generating considerably more energy off-axis in the presence region than those of the THOR. The audible result of this—especially in rooms that are somewhat live—will be an increased perception of upper midrange/lower treble energy. ❖

John Stone
SEAS USA

EDITOR'S NOTE:
A careful measure of the ports in the Froy cabinets confirms that they are exactly as specified, one-half inch between the two panels, contrary to SEAS manager John Stone's comment.

A NOTE ON TESTING:
The Froy (Mk3)s were tested in the laboratories of Audio and Acoustics, Ltd. using the MLSSA and CLIO PC-based acoustic data acquisition and analysis systems. Acoustic data was measured with an ACO 7016 ¼″ laboratory-grade condenser microphone and a custom-designed wideband, low-noise preamp. Polar response tests were performed with a computer-controlled OUTLINE turntable on loan from the Old Colony Division of Audio Amateur Corporation.

REFERENCES

1. J. D'Appolito, *Testing Loudspeakers*, Audio Amateur Corporation, Peterborough, NH, 1998.
2. J. D'Appolito, "THOR: A D'Appolito Transmission Line," *audioXpress*, May 2002, pp. 8–26.

NEW/OLD TECHNOLOGY

It is with some interest that I read your recent article, "Vacuum Tubes Born Again in Nanotube MEMS" (Nov. '02 *aX*, p. 46). It seems that it's an example of the old saw, "If it's in print, it must be true." I'd just like to point out a couple of instances of misleading or missing information, or the dreaded "proof by assertion."

Let's just look at the first sentence,

"Ever since the transistor was invented, engineers have lamented the slow, steady demise of the vacuum tube."

Really? This is a universal feeling among engineers? One can, with a statement like this, imagine the engineers at Intel sitting around their CAD stations, depressed because they really want to retool the upcoming Pentium-6 as an all-triode chip. And those poor engineers at Crystal, such a blue funk must rule their professional lives because they are unable to make their D/A and A/D converters using transistors. Why, they probably hold a weekly service, "In Lamentation of the Departed Hollow State." How much better a 256 times over-sampled A/D converter would be with tubes. Ah, the good old days.

Well, enough of that. Let's look at a statement a few paragraphs later:

". . . vacuum tubes simplify circuit design because they are true amplifiers all by themselves."

Part of this statement is true: yes, they are amplifying elements all by themselves. But the inference that the author apparently wants us to make—that transistors are not amplifying devices—is patently wrong. The author simply should know better and—writing for an industry electronic publication—should well know the difference between voltage-controlled current sources, current-controlled current sources, and the like, and should not assume one means "amplifier" and another does not. If, on the other hand, this is the view held by Mr. Zhu, the spokesman for the company, the author still should accept and exercise the responsibility for accurate reporting of technical points in a technical journal.

Another technical assertion made by Mr. Zhu is, well, curious:

"Even today, our transistor circuits are only about 5 percent efficient."
". . . engineers could downsize their solid-state equipment into smaller vacuum-tube-inspired designs that are 20 percent efficient."

This is patently and absurdly incorrect. I would challenge Mr. Zhu to directly produce evidence to support this assertion.

Assuming that Mr. Zhu is using any conventionally accepted definition of the word "efficiency"—that is, the ratio of power out to power in—he has at once completely contradicted the existence of millions upon millions of both transistor and tube amplifiers biased into class AB operation that are running at 40% efficiency. And his statement further ignores the fact that vacuum tube circuitry is not well suited for low impedance applications, and thus incurs the additional efficiency losses of prerequisite impedance matching devices such as transformers, not to mention the further filament power requirements. (Ignoring those last elements, there's not the slightest bit of data—be it theoretical or empirical—to suggest that there are any substantive differences in the efficiencies of two otherwise similarly biased amplifiers, one using solid state, one using vacuum tube technology.)

Now, there are certainly reasons for using vacuum-tube implementations that are driven by technological requirements. For example, many of the high-power RF components used in space vehicles are vacuum-tube-based, simply because they are far more resistant to the rigors imposed by the high-radiation environment of space.

But "efficiency?" Unless Mr. Zhu has coopted the definition of efficiency to mean something heretofore unheard of, his assertion quoted in the article is wrong.

As I mentioned at the outset, these assertions will, quite unfortunately, end up taking on a life of their own, whether they are technically valid or not. In an industry that already has more than its fair share of hokum, legend, and mythology, more of the same is needed like yet another hole in audio's too-porous head.

Dick Pierce
Hanover, Mass.

HIGH GM POWER TUBES

Since I have an adequate supply of high gm tubes obtained at 50 cents per from AES tube sale flyers, I can let the ignorant in on my little secret. 6KV8s and their ilk are generally high-mu triode-pentode combos; with the pentode being the interesting portion.

Also, of course, they are high-quality American tubes manufactured by RCA, Sylvania, or GE. These frame-grid beauties were designed as IF strip amps for TVs, so they are very linear and provide 20k µmho gm at 40mA. Power dissipation on the pentode section is around 3–4W. (Yes, I prefer to use the original American nomenclature as seen in the tube manuals, thank you very much!)

These tubes are very linear, and they make great cathode-follower drivers for MOSFET outputs (output impedance of 50Ω can drive 4–5000pF quite nicely!) They also work very well as power grid tube drivers in Class A2 and AB2 applications, such as enhanced triode sweep tube amps, or SV572-10 SE amps. This is because the high gm limits the distortion caused by the current draw of the grids.

David Wolze
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ANTENNA REVIEWS

I received the November issue and saw an article on AM radio antennas ("Tuning in on the AM Band," p. 32). A magazine about audio electronics construction projects will surely have an interesting perspective on AM antennas, I thought. Imagine my surprise: it was a review of two commercial projects, and it concluded that they worked.

Now, a brief Internet search will reveal lots of AM antenna construction projects. I did that search a year or so ago and printed out a few interesting designs. Surely one of the web folks might be interested in writing up the project for the magazine, or perhaps one of your regular authors could gather a few promising approaches and add some comments about AM propagation and reception theory.

That would be an approach worthy of *audioXpress*. I've subscribed since the very early days of *Audio Amateur* and still have all the issues. I expect how-to articles, not simple evaluations of commercial products. The combined magazine is very pleasing, by the way. I enjoy the addition of tube and speaker projects without having to subscribe to more periodicals.

Chris Campbell
Traverse City, Mich.

WHY BOTHER WITH TUBES

Pete Millett wrote a fine article on the tube headphone amp (Nov. '02 *aX*). It is very detailed, practical, and thorough. Of course, the question that I, as a tube-head, would like to pose is "Why bother with the tubes?" The circuit has an IC operating Class B, and a solid-state load on the tubes.

First of all, you would have a much better result by replacing the IC with a single-ended power FET output stage. Just snip the IC out, short the bias TP to ground, and direct-couple the FET source to the coupling caps. Now, you have an all-Class-A headphone amp.

Of course, you would obtain an even better result by going with a common-source Class-A FET output stage and running about 10dB of negative FB from the drain to the cathode. And lose that current source IC!

I also seem to remember that there were low-voltage driver tubes used to drive the bases of the output trannies on those radios. I think that these would enable us to go all tube! In this case, the original question of "Why bother with the tubes" would be rendered moot!

David Wolze
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Pete Millett responds:

Whenever designers propose a hybrid tube/solid-state amplifier, they're always met with responses of "why bother with the tube," or "why bother with the (transistor, FET, IC, and so on). So, I expected to hear this about the hybrid headphone amp project.

Mr. Wolze seems to miss the point of this project entirely. The point of this circuit is to allow a hobbyist to safely and inexpensively experiment with, and listen to, the characteristic "single-ended tube" sound. It is not to build a low-distortion amp, a high-end amp, or any other kind of amp.

Even though the output stage is a Class-AB bipolar IC design (not Class-B as Mr. Wolze suggests), it does not by any means dominate the sound of the amplifier. Why? Because the distortion products generated in the tube input stage are on the order of 100x that of the output stage. You hear the harmonics of the tube circuit, and little else.

There are many other ways to implement a headphone amplifier. Mr. Wolze is correct that a single-ended FET output stage could be used in place of the BUF634. I considered this when I designed the amp, but decided that the BUF634 would be a better implementation for most people.

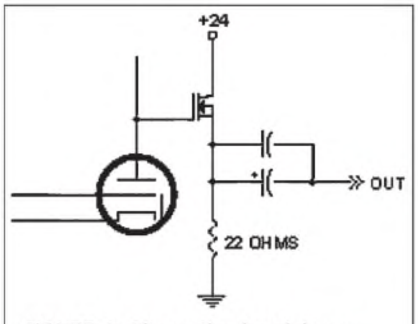

FIGURE 1: Alternative headphone amp circuit.

G-mill-1

If I understand what he proposes, you could change the output stage to a source-follower circuit as shown in Fig. 1. I doubt that you would get less distortion from this circuit, but you would remove any crossover distortion that might be generated in the

BUF634, as well as introduce more second harmonic distortion. By itself, I agree that the SE FET circuit would likely sound better than a BUF634—but with the tube in front of it, the differences you hear will be small.

There's also the small matter of bias current. If you kept 22Ω in the drain, the circuit would draw about 450mA per channel, and dissipate about 5W in the drain resistor, as well as in the FET. Better get bigger resistors, heatsinks, and a hefty power supply. And, by the way, you just added a bunch of money to the project!

To be fair, you could raise the resistor to something on the order of 200Ω and lower the current, and still drive most headphones. You would compromise the capability of the amp to drive a reasonable voltage into low-impedance headphones, though. A "much better result"? I doubt it, but feel free to try it, especially if you use only headphones with 300Ω or higher impedance.

As for adding 10dB of negative feedback around the circuit and changing to a common-source configuration, if you want to do that, forget the tube and just use an op-amp. You'll completely lose the harmonic profile that I was trying to get—basically, you will remove the sound of the tube. Would it sound better? Personally, I doubt whether I would like it, but maybe some people would.

The constant-current diode on the tube plate (it is not an IC, rather a discrete FET device internally configured as a CCS) was used to allow the tube to swing near the power supply with low distortion. Of course, you could use a resistor as a plate load—but you would need to raise the power-supply voltage and/or live with a restricted voltage swing and more distortion. As to whether a resistive plate load or a CCS sounds better, that's a matter of debate. I've been quite impressed with (god forbid!) solid-state CCS loads on the plate.

If you want an all-tube design, there are many possible implementations, including conventional designs that use transformer coupling from the plate, or circuits that use no output transformer (take a look at the output impedances obtained in the article about high Gm tubes!) Again, that's not the point—these circuits all cost more, and most use high voltages and/or global feedback that I was trying to avoid.

Realize that everything is a compromise—and keep an open mind!

VALVE ENTHUSIAST

I am nostalgic. Nostalgic for the good old days of *Glass Audio* magazine.

While reading some back issues of *GA*, I realized just what we valve enthusiasts lost when *GA* "married" your other publications. Take the October '02 issue of *audioXpress*, for example: just 2½ pages out of 72 are devoted to valve-related topics. We are not happy!

Nevertheless, I shall continue subscribing to *aX* for the foreseeable future, mainly so that I can purchase some of the excellent books your company reprints, even if they are very expensive over here—currently we pay $1.90 for $US1!

The book *Fundamentals of Radio-Valve Technique* is my current favorite. Marvellous stuff!

Terry Robinson
Victoria, Australia

Thanks for your comments and your kind words about Glass Audio. I think you might find another tube page in the October issue of aX if you look again. However, isn't there a hidden assumption in your accounting? Our magazine today is about audio. All of audio. Does your system work without signal cables? Isn't a report on cables relevant to your system? Isn't Martin Colloms' advice about rooms relevant to someone who uses tubes in the system? Do tube lovers never consider using a subwoofer?

Some readers have ventured out of playing an instrument with only one string, discovering the joys of building some devices which contain both tubes and op amps, or the delights of building a speaker worth much more than they could possibly afford and designed by some of the best in the business. Time to drop the blinders and look at the rest of what audiophiles are being offered.

All in all, we believe the marriage is a success and is about the whole subject of audio. If you check out a year of aX, you will find that our treatment of the three disciplines is about as balanced as we can make it. A teacher wouldn't pass or fail any student on the basis of one test or one term paper. –E.T.D.

THE AMP REVIEW

Regarding the review of the S-5 Electronics Amplifier in the Nov. '02 issue (p. 47), using four triode/pentode tubes originally designed for television vertical deflection amplification service to make a stereo, push-pull amplifier is very clever. As to pentode, push-pull amplifier tubes operating in Class-A or -A1, I find that generally output power is

equal to the allowable plate dissipation; i.e., 6L6s with allowable plate dissipation of 30W can produce 30W. The RCA tube manual covers triode/pentode tubes intended for vertical deflection amplification service: 5FV8 = 6FV8 and 10C8. These have allowable plate dissipation values of about 2.2W, so that by the relationship mentioned previously, only about 2.2W output can be expected, but 8W is claimed, and at 1000Hz, with a favorable load, 8W was achieved, but at a horrendous distortion level.

What I am leading up to is to suggest that, perhaps, the allowable plate dissipation is being exceeded, that this excessive heat is contributing to the sockets scorching and melting, and that this will result not only in melting sockets, but with very short tube life as well. This suggests the questions: Are replacement tubes readily available, where can they be obtained, and how much do they cost?

This amplifier has very high levels of frequency and amplitude distortion—this is objective fact. Duncan and Nancy's opinion that it nevertheless sounds pretty good is their subjective assessment. Is it fair to conclude that either they can't hear those distortion products, or that they hear them, but find them pleasing? I think that their recommendation of this amp can be ignored at least until other amps in this price range are also tested and auditioned.

David J. Meraner
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Duncan and Nancy MacArthur respond.

As we mentioned in the review, tube amplifiers sometimes measure poorly but receive good reviews for sound. This observation is unique neither to us nor to audioXpress.

"ELEPHANTS" CONTINUED

Paul E. Davis' letter (Nov. 2002 *aX*, p. 61) about audio "elephants" getting in the way of good speaker design prompts me to add a few more "elephants" (notes of caution) to his three. Elephant #0 should be "what the heck are you measuring?" In his case, he is measuring with an SPL meter in his room. This is OK in some rough sense, but the danger is that it combines the direct sound with the reflected sound, and your ears/brain perceive those differently.

At a recent Audio Engineering Society meeting in L.A., Jamie Anderson told a funny story about touring with the Grateful Dead back when FFTs were very new to pro-sound. They stuck a mike in the air, made a measurement, meticulously EQd every narrow peak and dip, and wondered why it sounded bad. The retrospectoscope revealed that they were measuring the effects of reflections mixed in with the direct sound.

In fact, they should have been EQing only to the direct sound's response. This is why speaker designers build anechoic chambers, or buy time-gating equipment, or employ one of Vance Dickason's favorite methods by "hauling everything outside," as Paul Davis puts it. If you can't measure without the reflections, you almost might as well not measure at all (except perhaps the bass region).

Elephant #4 I'd call "how good is your equipment": in this case, the Radio Shack SPL meter. I'm not sure about the current one, but my old one shows a large upper midrange response peak in the owner's manual, even in the "flat" mode!

In any case, newbies to speaker building should definitely "beware of elephants"—and if you can't measure your driver's actual impedance and quasi-anechoic frequency response, do yourself a favor and buy a kit.

Eric Guarin
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TUBE SOURCE

Where can I buy the Russian 6S17K-V tube mentioned in Eric Barbour's "A Planar-Triode Phono Preamp" (Nov. '01 *aX*, p. 18)? The supplier listed claims he never had the tubes and can't get them.

Richard P. Robinson
richmix@erols.com

Eric Barbour responds.

There is a new dealer of Russian tubes, Peter Zarytov, with a website: http://gstubes.com. He claims to have access to 6S17K-Vs, under the original number 6C17K-B.

If he can't help, I don't know what else to suggest. Many of the surplus Russian electronic dealers have gone out of business in the past year. All I could suggest is to write to New Sensor and ask them to import some of these tubes. Perhaps if enough people do so, they will take action.

VIVA LA RESISTANCE!

Double thanks for printing two of my letters in the Oct. issue—especially the one about the good, cheap MCM tweeter (pp. 62–63). I can imagine more than one of your advertisers being unhappy about seeing such a thing in your magazine—but you printed it anyway! Viva La Resistance!

And what is La Resistance? It's those who use and trust their ears. It's those who build their own stuff—not just to save money but because there is a kind of faith that there is something better: something you can't buy off the shelf.

It's my belief that this is why a mag like *Stereophile* is doomed. Their premise is if you spend more money, you can reach musical ecstasy (do I sound like the late Harvey Rosenberg?).

This is essentially not true. You're only going to get as much out of your system as you put into it. This isn't to say money isn't part of the equation—it is, but it's the consciousness of the builder that matters more. Magic isn't something you buy—it's something you do.

Rick Bergman
Missoula, Mont.

HORN FOR GUITAR

I have some questions concerning the DR10a horn by Bill Fitzmaurice (June 2002 *aX*, p. 16). Can I use the Eminence Beta 10 transducer in this horn? How is the compression chamber behind the loudspeaker calculated? Is this a bass-reflex system or a mixture of bass-reflex and horn? I hope you can answer these questions, because I am planning to build this horn for my guitar.

Sebastian Braun
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Bill Fitzmaurice responds:

The Beta 10 will work, since its parameters are well within the preferred limits. I assume that purchasing a Carvin PS10 from California is not the best route for you in Germany. But if Eminence products are available to you, I'd go with the Gamma 10, whose higher f_s will provide smoother response.

The woofer chamber, being vented, is not a compression chamber. The system is a combination of horn and bass reflex. The volume of the chamber was not calculated. My designs are 90% empirical, 10% formulaic. I choose a target cabinet volume, design the largest horn possible to fit inside of it, and what is left over becomes the rear chamber. This unorthodox method works because most of the performance is derived from the horn section, and the high f_s low Q MI drivers used are designed for relatively small enclosure volumes, anyway.

Your intention to use a DR10a for a guitar brings up questions from me. Is this for electric guitar? If so, the DR10a, or any horn-loaded cab, is probably not what you need. Classic electric guitar tones are heavily colored and distorted, and are best achieved with either very small infinite baffle cabinets or open back cabinets.

Acoustic guitar is another matter, because that instrument is best projected through a fairly clean wideband speaker, which the DR10a definitely is. However, the vented chamber of the DR10a extends bass response of the cabinet to well below the passband of the horn, which is great for electric bass, keyboard, or PA, but unnecessary for guitar. If you intend your DR10a strictly for acoustic guitar, you'll get better mid-bass response by not using any ducts, thus making the rear chamber a compression chamber. Response below 80Hz will suffer, but that matters not, since the guitar doesn't go down there anyway, and response around 125Hz will be better.

KIT WANTED

I've just read the November 2002 issue of *aX*, featuring a valve/solid-state headphone amp project ("Build a Low-Voltage Tube Hybrid Headphone/Line Amp," pp. 20–31). This is just what we have been looking for! Unfortunately, despite searching through some Australian electronics catalogs, some of the specialized parts seem to be unavailable in Australia. It would be good if this project could be offered as a kit.

Terry Robinson
Victoria, Australia

HELP WANTED

I recently found a product named Bozak model #N-10102A that appears to be a crossover for two tweeters (model #B-200Y) on a frame that held a 12″ woofer, which is missing. The two midrange/bass units are 5″ (model #B-209B). All drivers are 8Ω.

My question is what year was it manufactured, and what are the crossover frequency, the wattage of the speakers, and perhaps the cost of the product at the time? (The label reads R.T. Bozak MFG. CO., S. Norwalk Conn., USA.)

David Mozingo
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Ayden, NC 28513
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Readers with information on this topic are encouraged to respond directly to the letter writer at the address provided.–Eds.

Book Review

Master Handbook of Acoustics, Fourth Edition

Reviewed by Duncan MacArthur

Master Handbook of Acoustics, Fourth Edition; F. Alton Everest. McGraw-Hill, 2001, 615 pages, ISBN 0-07-136097-2, $34.95.

The word acoustics means different things to different people. To a stadium designer, acoustics is about ensuring clear vocal reproduction at every seat. Concert halls are usually smaller, but the designer must ensure that the entire musical spectrum is reproduced at each seat.

To the recording engineer, acoustics means the effects of the studio on the sound produced by musicians as well as the reproduction of sound in the control room. Audio enthusiasts may believe that the effects of smaller, often multipurpose, rooms on music reproduction are most important. All these and more are acoustics, with the result that any single book, even a lengthy one, cannot cover everything in detail.

ACOUSTICS

The detailed study of acoustics can also be mathematically complex. Fortunately, this level of detail, although critical to the professional acoustical engineer, is not essential to understand basic listening room acoustics. In his *Master Handbook of Acoustics*, Mr. Everest successfully keeps the mathematics to the absolute minimum required. He uses many graphs and tables, a few photographs, and a minimal number of formulas to describe various phenomena. Readers with mathematical backgrounds might prefer more detail, but even they will appreciate explanations that don't immediately resort to arcane mathematics.

In this volume Mr. Everest concentrates on the acoustics of small rooms, such as home listening rooms and studios. He also discusses the principles behind—and the effects of—various commercial acoustical tools as well as a number of "home-built" alternatives. Additional short chapters cover scientific fundamentals and advanced techniques. Finally, the author includes extensive references to further reading and primary source material.

In an attempt to give a flavor of a thick book in a short review, I have divided the chapters into four groups. In the early chapters, Everest covers general introductory material, the small amount of required mathematics, and several topics primarily of scientific interest. Next comes a group of chapters with some content useful to the home listener mixed with other material. A number of chapters deal explicitly with recording studios and techniques. Finally, six chapters covering absorption, diffusion, and listening room acoustics contain material highly useful to the audio enthusiast.

INTRODUCTORY MATERIAL (1, 2, 3, 4, 5, 11, AND 12)

The first five chapters contain introductory material required to understand the rest of the book. No prior knowledge of acoustics or higher mathematics is needed to understand these chapters. The more experienced reader may prefer to skip these chapters the first time through, but can refer to them at need if a more advanced topic doesn't make sense.

Chapter 1, "Fundamentals of Sound," introduces the basic concepts of sine waves, wave propagation, and the relationship between frequency and wavelength. For the first time (of many throughout the book) graphical explanations supplement the mathematical formulas. The concepts of harmonics, octaves, phase, and spectrum are introduced, and this chapter concludes with a short description of electrical analogies.

Since the response of the human ear is intrinsically logarithmic, some discussion of logarithms, exponentials, and scientific notation is required even in a fundamentally non-mathematical book. This material, along with the concept of acoustic power and numerous examples, is covered in Chapter 2, "Sound Levels and the Decibel."

The concepts of subjective loudness as well as the relationships between loudness and frequency and pitch and frequency are introduced in Chapter 3, "The Ear and the Perception of Sound." This chapter introduces Fletcher/Munson-type response curves and explains the function of a loudness (as opposed to volume) control. The description of the precedence or Haas effect is particularly important for future discussions of loudspeaker placement.

Chapters 4 and 5, "Sound Waves in the Free Field" and "Speech, Music and Noise," touch briefly on free-field relationships and sources of sounds. I was particularly glad to see that Everest has included two charts to which I refer frequently: "Power of Musical Sources" (Table 5.1) and "The Audible Frequency Range of Various Musical Instruments . . ." (Fig. 5–11). He describes the distinction between "white" and "pink" noise as well as the concept of musical distortion.

The concepts of "Diffraction" and "Refraction" of sound introduced in Chapters 11 and 12 are primarily of scientific interest. Other than the introduction to diffraction around speaker cabinets in Chapter 11, this material appears to be included for completeness and is not necessary to understanding the remainder of the book.

CHAPTERS OF MORE INTEREST (6, 7, 8, 10, 24, 25, 26, 27, AND 28)

A number of chapters in the *Master Handbook of Acoustics* contain some useful material along with material that is not directly applicable to home listening. The usefulness of this group of chapters depends greatly on the reader's personal definition of acoustics.

Chapter 6, "Analog and Digital Signal Processing," provides a very brief introduction to a topic that probably will be a key development in the next decade. At this time, signal processing (especially digital) offers a taste (albeit an expensive one) of good things to come.

Reverberation, as discussed in Chapter 7, is one of the key ingredients in concert hall–and other acoustic space–design. Understanding the concepts of reverberation decay time, variations with frequency and position, and the Sabine equation is important for hall design. Practical advice on these subjects is reserved for later chapters.

Control of externally generated noise–whether from the neighbor's lawnmower or from your own HVAC system–is a constant concern for many music listeners. For most of us the most common "solutions" are to "grin and bear it" or "listen some other time." In Chapter 8, Everest provides some ideas for improving this situation, but not a detailed discussion.

Chapter 10, "Reflection of Sound," gives some information on reflection of sound waves off various shapes. Many of the distortions generated by these reflections are discussed in Chapter 25. Although Chapter 10 includes many examples of problems, solutions are mostly left for later chapters.

Adjustable acoustics is potentially important to the home listener, especially when the listening room does double duty as living room or family room; however, Chapter 24 is oriented more towards recording studios than towards listening rooms. This chapter also reads a bit like an advertisement for ASC and RPG products.

As with Chapter 6, Chapters 26, 27, and 28 offer tantalizing glimpses of advanced technologies. These chapters discuss "Room Acoustics Measurement Software," "Room Optimization," and "Desktop Auralization," respectively. As you might expect in an overview treatment, all three chapters consist mostly of glowing descriptions of single commercial products. Rather than viewing this as advertising, I interpret these as examples of very impressive technologies that are becoming available (for a price). These technologies are currently more in the realm of acoustic professionals than hobbyists.

RECORDING ORIENTED (17, 20, 21, 22, AND 23)

A number of chapters in the *Master Handbook of Acoustics* are specifically aimed at recording studio design and technique. Since the acoustics of studios and living rooms are related, some elements of these chapters may be of interest to the home listener. In particular, many of the photographs, although specifically of studio applications, will also interest the home listener.

Chapter 17, "Comb Filter Effects," discusses the effect of the interference between two sound sources. Although this is primarily microphone oriented, Mr. Everest also discusses interference between two stereo speakers and between drivers in individual speaker systems.

"Quiet Air for the Studio," as discussed in Chapter 18, is a practical continuation of the earlier chapter (8) on noise. Many of the techniques are potentially applicable to home listening rooms; however, few of us have the opportunity to design and build a room specifically for listening.

Chapters 20 and 21, "Acoustics of the Small Recording Studio" and "Acoustics of the Control Room," would be more interesting if they didn't immediately follow Chapter 19, which discusses acoustics of the listening room. As it is, Chapter 20 covers much of the same material from a recording studio point of view with more emphasis on diffusion and noise control. Chapter 21 discusses the time delay gap (first mentioned in Chapter 3) as well as LEDE (Live End Dead End) techniques and more uses of RPG products. Everest includes numerous photographs of actual implementations.

The chapter on "Acoustics for Multi-Track Recording" (#22) will be of interest to studio designers. Chapter 23, "Audio/Video Tech Room and Voice Over Recording," contains another discussion of room modes (the most complete is in Chapters 15 and 19) and continues the discussion of Chapters 20 and 21.

MOST USEFUL FOR HOME LISTENING (9, 13, 14, 15, 16, AND 19)

I've saved the best for last. The *Master Handbook of Acoustics* contains six chapters that are indispensable for understanding listening room problems, potential solutions, and commercial implementations of these solutions. The interaction between the listening room and sound reproduction is often misunderstood. Among other things, these chapters explain why different rooms sound different and what you can do about it. Even if you have no interest in building room treatments, this material helps immeasurably in understanding the commercial products that are available.

In Chapter 9, "Absorption of Sound," the author covers a range of topics including evaluation of absorption, sheet absorbers (e.g., foam, fiberglass, drapes, and carpet), and bass absorbers such as traps, diaphragms, and Helmholtz resonators. This chapter is also the place to look for a thorough description of polycylindrical absorbers or "polys." In addition to containing an enormous amount of DIY information, this chapter provides an extremely useful explanation of the effects of commercial products. I would buy the *Master Handbook of Acoustics* for this chapter alone.

Room modes and their relationship to room dimensions are one of the most important acoustic issues for the home listener. Chapter 13, somewhat confusingly titled "Diffusion of Sound," addresses these questions as well as the effects of room shape and general sound field issues. "Modal Resonances in Enclosed Spaces" are revisited in Chapter 15 with emphasis on calculating modes in rectangular rooms.

In Chapter 19, Everest discusses room modes yet again with emphasis on solutions and bass traps. Chapter 19 also includes a discussion of reflections and midrange clarity. Although the material on room modes is spread among three chapters (and more), the complete discussion is quite thorough. This set of three chapters is an excellent introduction to the concept of room modes.

One significant advance in acoustics in the last couple of decades has been the application of diffraction gratings to sound waves. Chapter 14, "The Schroeder Diffusor," traces the history of grating-based diffusors from the original Schroeder models through the more recent quadratic residue and primitive root designs. This chapter concludes with a huge amount of design information, but the description of commercial systems is somewhat "RPG-centric."

Finally, in Chapter 16, "Sound Reflections in Enclosed Spaces," the discussion moves away from bass characteristics of listening rooms to reflection effects in the midrange. Topics covered here include echoes, imaging, and spaciousness.

CONCLUSION

If you're looking for a single introduction to acoustics that doesn't require much mathematics, it's hard to see how you could go wrong with the *Master Handbook of Acoustics*. As I mentioned earlier, its very breadth of coverage means that the coverage on any given subject may not be deep. In such cases, the references to other material may be more important than the text itself. However, I highly recommend the *Master Handbook* for basic explanations of numerous acoustic effects and techniques. This new, larger edition is a worthwhile addition to your library even if (or perhaps especially if) you already own an earlier edition. ❖

Audio Aid

A 1¢ Diffraction Detector

No matter how well you plan your speaker design, you never really close in on your desired results without testing and measurement. Here's a test you can use that doesn't even require the use of dangerous electricity.

When you listen to pink noise over loudspeakers, you'll often hear a "voice" that sounds like a resonance. In the literature, these resonances are often corrected with electrical filters—resistors, capacitors, and inductors in the crossover network, or with active equalizers. I found a different way to attack some of these resonances. If this method has ever been published, I've never read it.

When I was working with an old Audax HIF13J mid-woofer from the mid-1970s, pink noise sounded as though it was coming from a cat's mouth. I found and corrected the problem using an extremely inexpensive testing device—a plastic soda straw, free with your drink at your local fast-food restaurant.

Try this. Before you even go to get your straw, blow on the palm of your hand. Then blow on your fingertips, and then on the edges of the magazine pages. If you're wearing a shirt, blow on that. They all sound different, don't they? The difference is diffraction and turbulence.

The next step is to take your soda straw and go on a search and destroy mission. Blow on the face of your speaker driver, the surround, gasket, and basket. Blow on the edge between the driver and baffle. Blow on the cabinet corners—even rounded ones. Every time you hear turbulence, there's an opportunity to improve your sound.

I found with the old Audax driver that blowing on the square-profile cork gasket produced a rushing sound that had the same cat's mouth character as the pink noise. I molded some modeling clay around the gasket to smooth the profile, and found that I had also smoothed the sound. The modeling clay soon fell off, but I could always apply this solution to correct the diffraction noise in a more permanent material.

Granted, minimizing diffraction is nothing new. It was all the rage a generation ago, and since then many drivers have smooth, non-diffracting edges. You now find even budget systems with drivers mounted in molded baffles with smooth transitions or shallow horn-like exit profiles.

I don't know what methods the manufacturers use to find and correct diffraction, but this is an easy and inexpensive tool to help you do the same thing. You can find the causes of diffraction noise, and try out different baffle shapes in soft moldable materials before you take the time and expense of sculpting a baffle mold or investing in router bits that may or may not give you the results you're hoping for. You can also test different fabrics, foams, and other absorbing baffle treatments while you're still in the fabric store.

All you have to do is blow. ❖

Mark Hotchkiss
Mabelvale, Ark.

Classifieds

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Audio Aid

Current-Regulated Heater Supplies

By Michael Kornacker

In my article, "Direct-Coupled Circuits Need Regulated DC Heaters" (March '01 *aX*, p. 86), I discussed the need to regulate the heater voltage supply in directly coupled DC tube amplifiers to ensure stable operation. I would like to make some additional comments and talk about regulator circuits for tube heaters in regards to current supply.

CURRENT VS VOLTAGE REGULATION

Truth be told, a constant applied voltage, as in a regulated supply to a tube's heaters, does not guarantee a constant current drawn by those heaters. Because electron emission between the cathode and the plate of a tube is more closely tied with the thermal effect of electron current flow in the heater, a heater current regulator would be more efficient at controlling the operating point than heater voltage regulation.[1]

Although a voltage regulated supply will be superior to an unregulated supply in this respect, best results would be obtained with current regulation. With voltage regulation, any resistance, such as too small a hook-up wire, corroded tube pins, cold solder connections, or bad socket contacts, will cause voltage drops, and the heaters will not receive their fair share of current. With current regulation, after the current is set it will remain fixed and all these detrimental factors become irrelevant.[2]

Current regulation also adds longer heater life due to its soft-start character. In a normal situation, when power is applied, a cold heater acts like a short circuit drawing a huge in-rush current flow before it heats up. Every time you turn it on, the accumulation of the filament's transitions from ambient to white-hot eventually weakens it. Do it one too many times, and the tube finally burns out. A current regulator will limit this current spike, even when cold, to the heater's normal steady-state current draw and as a result prolong tube life.[3]

When an incandescent lamp is turned on (which is basically the same as the heater in a vacuum tube), its filament can draw up to 18 times its normal operating current. Even after 5ms, current flow could still be up to five times normal. So, as the filament heats and light is emitted, its resistance increases and the peak current decays "quasi-exponentially" to its normal rate.[4]

You have seen that room light bulbs can burn out with a flash when you first flip on the wall switch, usually never after the lamp has been on for a while. This shows that the high mortality rate of lamps—and, thus, tube heaters—is caused by being constantly overstressed at turn-on. Current regulation would eliminate this problem and save money.

THE CIRCUIT

With today's technology, implementing current regulation is as easy as voltage regulation. *Figure 1* is an LM317 regulator chip configured as a current regulator. The input voltage to the LM317 must be greater than the required output voltage for the load by about one and a half times. The current limiting resistor R is equal to 1.25 divided by the total load current of the heaters. The wattage of R should be greater than the product of the above quantities.

When you power up the regulator circuit, check for both the required current through the heaters and for the voltage across the heaters. You may need to fine tune R to obtain the proper values because the resistance R calculated is rarely the actual value required.

In my own amplifier ("Eico HF-86," *GA* 4/99, p. 68), I have two 12DW7s rewired for 12.6V heaters, which draw a total of .3A. That makes R equal to 4.17Ω at 2W. The amplifier's transformer has two 6.3V secondaries—one for the HV rectifier tube, which was unused since I replaced the tube with semiconductor diodes, and the other for the rest of the tubes (the two 12DW7s and four power output tubes).

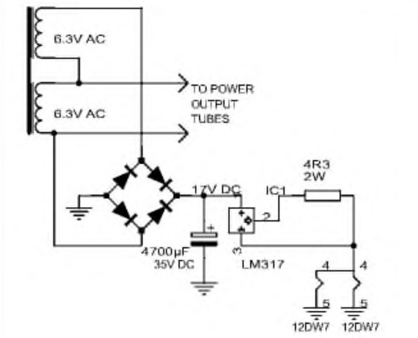

FIGURE 1: The LM317 current regulator circuit used for the preamp tubes of the amplifier.

G-1960-1

I separated the 12DW7 heaters from the rest so that I could power them with DC. To obtain the required DC voltage, I connected the two secondaries in series and full-wave-bridge-rectified and filtered it to obtain about 17V. I connected this to the LM317 current regulator, and then to the heaters. I also needed to heatsink the LM317.

After adjusting the value of R for 12.6V and .3A at the heaters, everything worked beautifully. I ended up using a 4.3Ω 2W (meas. 4.4Ω) in parallel with a 62Ω half watt (meas. 66Ω). To my surprise, I found that there was no residual ripple voltage across the heaters as there was across the filter cap. Apparently the current regulator removes the ripple just like its voltage regulator counterpart.

Using a heater current regulator should provide the utmost in stable operation of vacuum tubes, regardless of whether it is direct coupled or not. This will also have the benefit of an automatic soft-start and reduce the possibility of hum being induced into the cathode circuit and the rest of your amplifier. ❖

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1. *Regulated Power Supplies*, Irving M. Gottlieb, 4th Edition, Tab Books, 1992, p. 15.

2. Ibid., p. 40.

3. Ibid., p. 41.

4. "MOSPOWER Application Notes," Siliconix Inc., Rudy Severns, Editor in Chief, Santa Clara, Calif., 1984, p. 6–82, 83.