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Dr. Joseph D'Appolito has been working as consultant for Usher Audio since early 2001. A world renown authority in audio and acoustics, Dr. D'Appolito holds BEE, SMEE, EE and Ph.D. degrees from RPI, MIT and the University of Massachusetts, and has published over 30 journal and conference papers. His most popular and influential brain child, however, has to be the MTM loudspeaker geometry, commonly known as the "D'Appolito Configuration," which is now used by dozens of manufacturers throughout Europe and North America.

Dr. D'Appolito designs crossover, speaker cabinet design, and tests prototype drivers for Usher Audio, all from his private lab in Boulder, Colorado. Although consulting to a couple of other companies, Dr. D'Appolito especially enjoys working with Usher Audio and always finds the tremendous value Usher Audio products represent a delightful surprise in today's High End audio world.

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

JOHN STUART MILL

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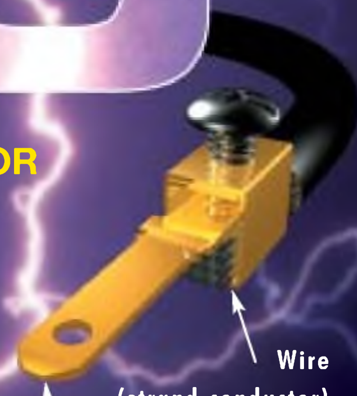

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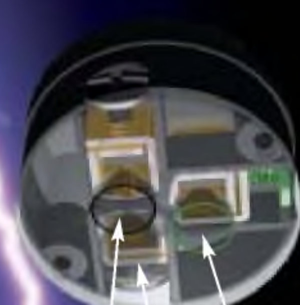

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A Simple, High Quality AM Tube Receiver

You, too, can re-discover how good AM radio can sound when it is done right. **By Scott K. Reynolds**

This construction project began when I read a letter from Vince Roberts in the February 2001 issue.[1] Mr. Roberts requested an article on an AM tuner using vacuum tubes, and reading his letter inspired me to build one. I tried to come up with a design that would be easy for the home constructor to build and also give good audio quality. Both of these goals meant that I had to keep the radio frequency (RF) circuits as simple as possible, but a large, efficient antenna more than compensated for the low RF gain and resulted in a receiver that has surprisingly good sensitivity.

My wife was at first skeptical of this project, but in the end she liked the radio so much that I had to move it from my workshop into the study. My wife is a National Public Radio (NPR) enthusiast, and we live in a fringe reception area where the local NPR AM station (WNYC) usually comes in with a fair amount of static and noise. To even my surprise, this new AM receiver brings in WNYC with consistently lower noise than any other radio in the house, and it has a rich and detailed sound the other radios lack. It made my

PHOTO 1: Front view of the radio receiver. The RF circuits are on the left, with tubes V1 and V2 at the left rear.

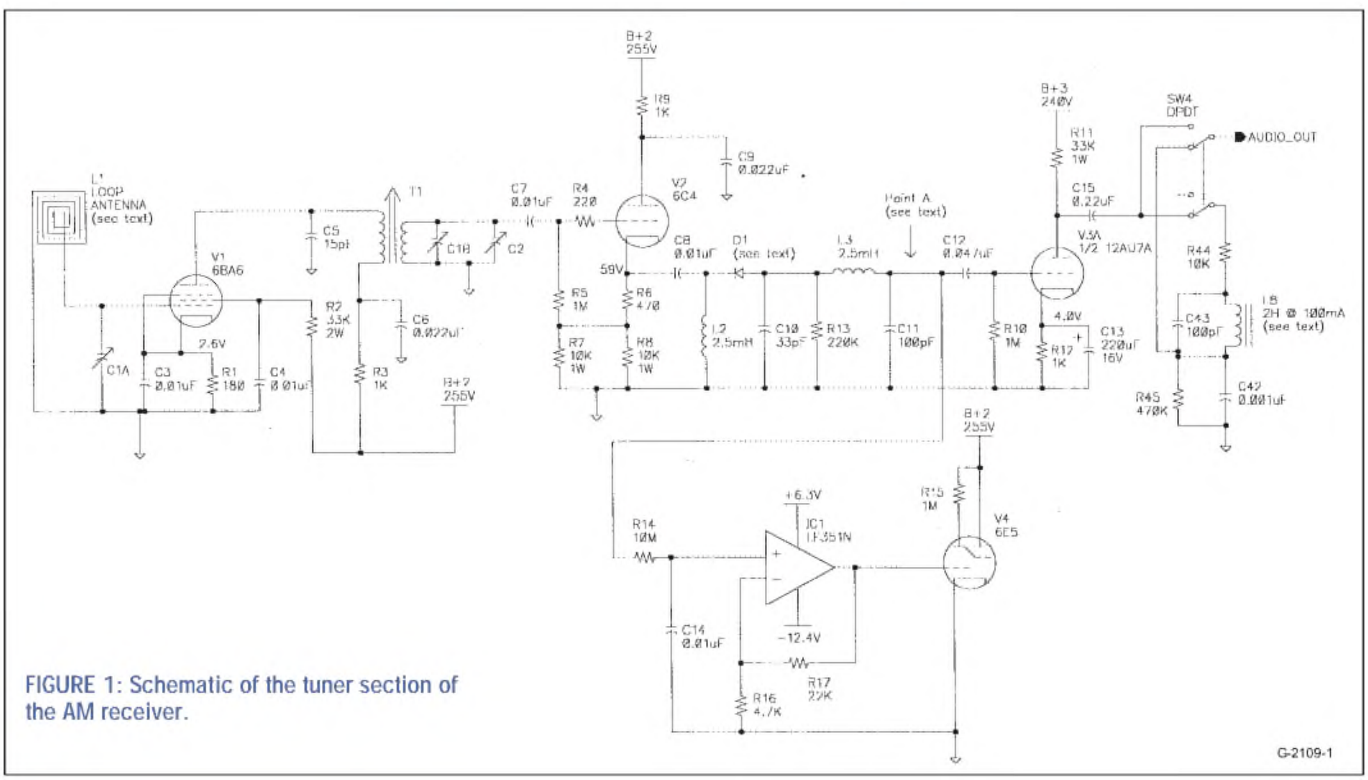


FIGURE 1: Schematic of the tuner section of the AM receiver.

wife appreciate (and me remember) how good AM radio can sound when it is done right.

BACKGROUND

When I read Mr. Roberts' letter, I thought about what sort of receiver would be best for home construction. I eventually settled on a tuned radio frequency (TRF) receiver with a single stage of RF amplification (*Fig. 1*). There are only two tuned circuits in this receiver, so construction and alignment are easy, as I will describe later. So that the trade-offs I made in arriving at this design will be clear, I'll first discuss some other receiver circuits that I considered but did not pursue.

The simplest receiver capable of giving satisfactory results in everyday operation is undoubtedly a regenerative detector followed by audio amplification. In its simplest form—as it was first built in the 1920s—the regenerative detector circuit consists of a triode grid-leak detector with positive feedback around it. The positive feedback greatly increases the gain of the stage and the selectivity of the tuned circuit. Unfortunately, it also increases the distortion and noise and makes the detector prone to oscillation.

These simple regenerative circuits can work amazingly well at pulling in distant signals, but their poor audio quality makes them better suited to experimentation than pleasant listening. The early editions of the *Radio Amateur's Handbook* described these regenerative receivers in detail, but the more recent editions discuss them only in passing.[2] Antique Electronic Supply has books on regenerative receivers if you would like to experiment with them.

Most radio receivers are superheterodyne[3] circuits in which the incoming RF signal is first converted to an intermediate frequency (IF) signal before being amplified and detected. Superheterodyne circuits are popular because they can have high gain and selectivity while also being stable and easy for you to tune.[4]

Superhets need to be carefully designed to give good audio quality for AM signals, however. If the IF transformers are too highly selective, the high-frequency components of the received signal can be attenuated. Also, the frequency conversion process can generate high intermodulation distortion.

When I considered building a superheterodyne receiver and writing about it for *audioXpress*, I was concerned

PHOTO 2: Rear view of the radio receiver.

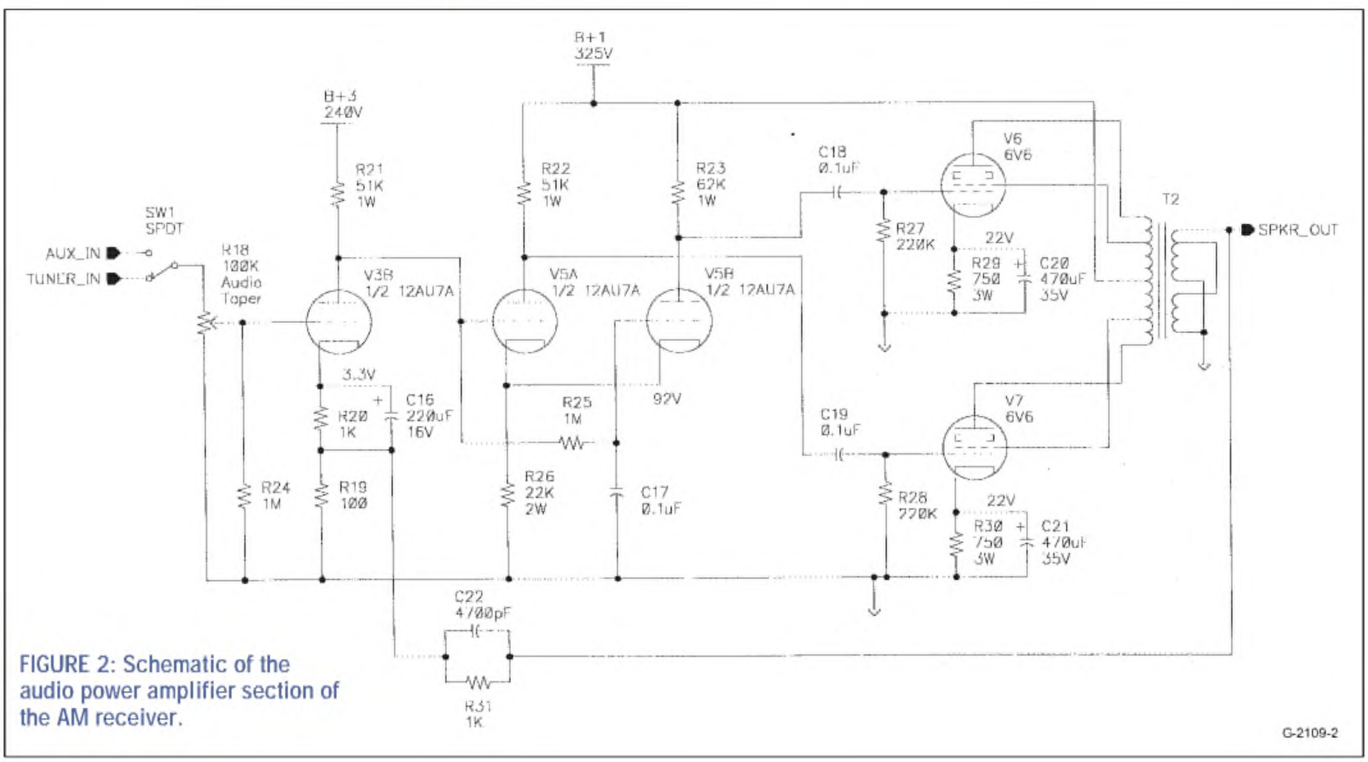


FIGURE 2: Schematic of the audio power amplifier section of the AM receiver.

about the complexity of the RF circuitry and the difficulty that the home constructor might have in aligning a newly built radio (aligning several stages in a newly built superheterodyne radio is more confusing than re-aligning an already working radio). I was also concerned about the difficulty of finding parts, especially the tube-compatible IF transformers, since I don't think they are being manufactured anymore. Antique Electronic Supply has a few surplus IF transformers, but it's difficult to know how long they'll be available. I considered winding my own transformers, but I decided to follow Mr. Roberts' advice to "start simple."

PHOTO 3: Distant rear view of the radio, showing the construction of the antenna L1.

A TRF receiver offered a simpler alternative . . . if I could get the required RF gain in a single stage. Multiple-stage TRF receivers aren't so simple anymore, because of the problem of aligning multiple-tuned stages and having them track together over the band as the receiver is tuned. A high gain, multi-stage TRF receiver can be unstable due to unintended feedback.

Finally, TRF receivers suffer from variations in gain and selectivity as you tune from one end of the band to another, a problem which becomes worse as the number of stages increases. Back in the 1920s and '30s, designers came up with elaborate solutions to these problems, but I didn't want anything elaborate.

ANTENNA REQUIREMENTS

Having decided on a single-stage TRF design, I wanted to get the most performance I could out of it, and the antenna (component L1 in *Fig. 1*) seemed like the best place to start. I decided to build the largest loop antenna that would still be practical to receive as much RF signal as I could (without setting up an outdoor antenna). An amateur can readily build a better antenna than a manufacturer can, not having the same sort of space, shipping, or manufacturing limitations.

The RF signal received by a loop antenna is proportional to both the number of turns and the area, so it would seem that you want a large loop area and lots of turns. But in order for the antenna to resonate with the 365pF tuning capacitor C1A over the entire AM broadcast band (540–1600kHz), you are constrained to make the loop's inductance 238µH. The loop's inductance is approximately proportional to the square of the number of turns and the square root of the loop area, so selecting the antenna size gave me an interesting mathematical and experimental problem to solve.

I settled on a 24″ × 24″ (61cm × 61cm) square wooden frame, with nine turns of AWG 22 enameled magnet wire wound around the outside, with the turns spaced ⅛″ (3.18mm) apart. The spacing is important in order to minimize the self-capacitance of the antenna. If the self-capacitance is too high, the antenna cannot be tuned over the entire AM band.

My radio tunes from 560 to 1570kHz, which should be adequate for most readers. But some of you may wish to receive a station in the extended AM band from 1600 to 1700kHz (these are mostly low-power local stations, the most interesting kind). To do that, you'll need to take a turn (or half a turn) off L1, which may prevent you from getting stations below 600kHz. You can also tap L1 to make changing the number of turns easy.

For the antenna to tune the entire 540–1700kHz extended AM band without changing taps appears to require a smaller loop area, but, of course, that results in a less efficient antenna and a less sensitive receiver. The size of the antenna I chose is just one of those inevitable engineering compromises. I'll give more details on antenna construction and setup later.

CIRCUIT DESCRIPTION

The main tuning capacitor C1 is a dual-ganged 365pF air-variable capacitor, which used to be standard in AM radios. But these air-variable capacitors are becoming harder to find. RF Parts Co. and Antique Electronic Supply both have suitable parts. The fine-tuning capacitor C2, which is a 33pF air variable, is more readily available. But it can be tough to find one with a shaft for a knob (most

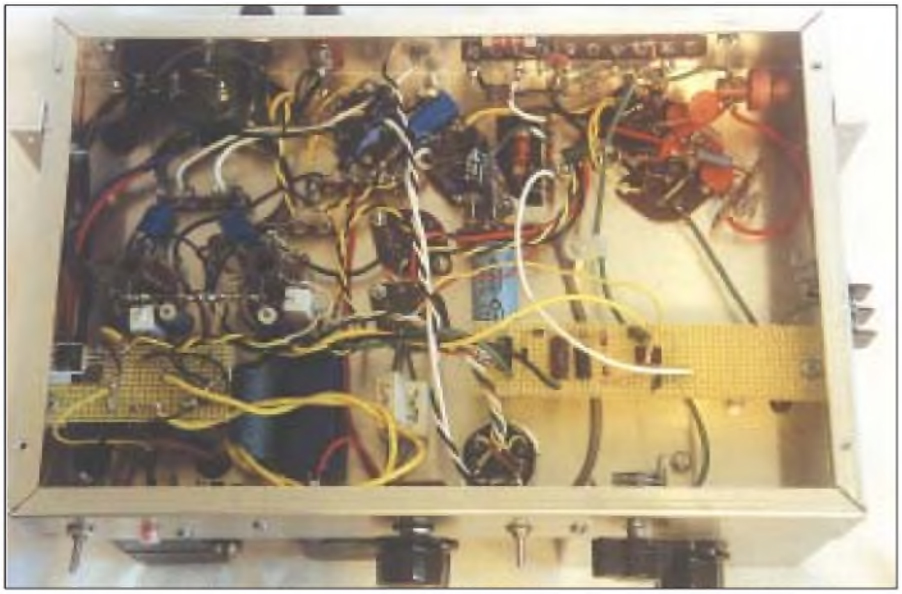


PHOTO 4: Underside view of the chassis. The RF wiring is at the right rear.

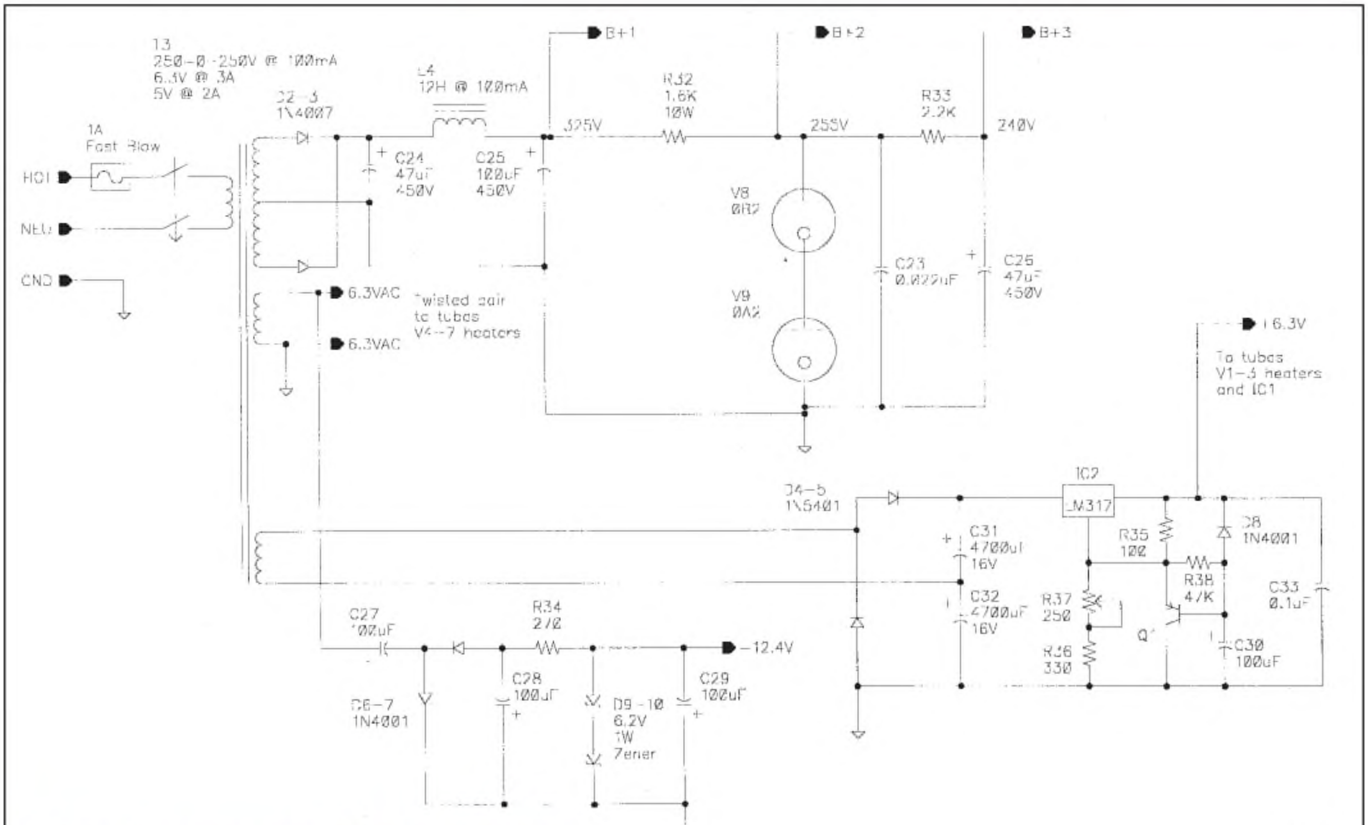


FIGURE 3: Power-supply schematic for the AM receiver.

have slots for screwdriver adjustment). Mine came from RF Parts Co.

All of the RF gain in the receiver (20–30dB) comes from remote-cutoff pentode V1, a 6BA6 or 5749/6BA6W. The quiescent cathode current in V1 is set to a relatively high 14.4mA by the value of R1. The high bias current is used to maximize the transconductance of V1, giving high gain and low noise. Raising the value of R1 lowers the transconductance and the gain, so you could achieve RF gain control by inserting a variable resistor in series with R1. I'll give more details on this later.

The output of V1 is transformer coupled to the detector circuit by T1, a slug-tuned RF transformer. T1 is Antique Electronic Supply part number P-C70-RF (I'm not aware that these RF transformers are available from any other source, but they aren't surplus, so they should continue to be available).

T1 is designed so that the primary circuit should resonate at a frequency below the AM band, but if the associated circuit does not have enough capacitance, some additional capacitance needs to be added to reduce the primary resonant frequency. That is the purpose of C5, a 15pF dipped mica capacitor. The secondary of T1 resonates with C1B and C2, forming a second tuned circuit at the same frequency as the L1-C1A combination. Because of its large physical size, the antenna L1 has more self-capacitance than the secondary of T1, so you need to add the additional fine-tuning capacitor C2 in order to make the two tuned circuits track properly over the AM band.

The detector circuit is a classic diode envelope detector used in countless AM radios. To get high sensitivity, use a 1N34A (or 1N60A) germanium point-contact diode (D1), which is often used in crystal radios and is 10–100 times more sensitive than a vacuum tube diode. These germanium diodes also have the advantage of being more linear than a vacuum tube diode at the low RF signal levels found in this receiver. Note that a standard silicon diode will not work for D1.

A cathode follower (V2) isolates the diode detector from the second tuned circuit, thereby preventing the diode from loading the tuned circuit and preserving high selectivity. Tube V2 (a 6C4 or 6100/6C4WA) is biased at a fairly high 10.8mA to give high transconductance. A cathode follower can be unstable at high frequencies when it has a capacitive load and a tuned circuit at the grid, which is the situation in *Fig. 1*. Grid stopper R4 prevents any such instability. You should install it immediately adjacent to the tube socket.

The output of the diode detector goes through a low-pass pi-section filter consisting of C10, L3, and C11, which removes the residual RF carrier. This filter has an audio bandwidth of about 6.6kHz, which is adequate for AM signals. You can increase the audio bandwidth by reducing the values of C10 and C11, if you want to experiment. With the RF carrier removed, the remaining signal at point "A" on the schematic (*Fig. 1*) consists of the audio signal and a negative DC component (there is also some second harmonic distortion). Capacitor C12 blocks the DC component and passes the audio to V3A, the first audio amplifier stage.

The amplitude of the DC component is proportional to the strength of the RF signal, which makes it useful for operating a tuning indicator. In my area, the strongest RF signals produce about −1.5V at point "A." But the 6E5 tuning indicator tube that Mr. Roberts requested requires about −8V for complete eye closure.

Therefore, the DC component is amplified by IC1, an LF351N JFET input op amp configured for a non-inverting gain of 5.6. An op amp with a JFET input is essential in order to provide high enough input impedance so as not to load the diode detector circuit. A standard 741 op amp will not work here.

I think the 6E5 tube is quite attractive, but it is becoming rare and expensive. You could also use a 10V full-scale analog meter movement at the output of IC1, or a 1mA meter with a 10kΩ series resistor.

Keep in mind that the output of IC1 swings from zero to about −8V, so wire the meter with the correct polarity. You cannot wire an analog meter directly to point "A" on the schematic because it will load the diode detector. Digital meters will not work for tuning because their response is not continuous. Some sort of tuning indication is essential for both alignment and normal operation, so don't be tempted to omit it.

AUDIO FILTER

Components R44–R45, L8, and C42–C43 in *Fig. 1* make up a passive low-pass filter that I added to the output of the tuner to eliminate (or reduce) the annoying 10kHz squeal that AM radios sometimes produce in an environment crowded with many stations. This 10kHz squeal arises because AM stations are spaced 10kHz apart, and along with the desired station you may also be receiving a weak, distant station 10kHz higher or lower in frequency (other local stations are usually 30kHz apart). The two signals "beat" together in the receiver to produce a 10kHz tone, and the only solution is to filter it out. This problem is less noticeable on a modern superheterodyne receiver because the IF filters are so highly selective, giving the receiver less bandwidth.

The filter has a −3dB point at approximately 5.5kHz, and it is down more than 15dB at 10kHz. A tiny filter choke (L8) acts as an audio choke to produce an LC filter. Capacitor C43 resonates with L8 at 10kHz to produce a sharp cutoff characteristic. The filter is designed to work best with load impedances between 50kΩ and 100kΩ (here it sees 470kΩ in parallel with the 100kΩ potentiometer R18, or about 82kΩ).

I don't recommend replacing L8 (a Hammond part #154M), because other filter chokes may have different audio characteristics. I added a switch (SW4) to bypass the filter when it isn't needed, since I didn't want to unnecessarily reduce the wide bandwidth that my receiver has. You can build the tuner without the filter by eliminating R44–R45, L8, C42–C43, and SW4, taking the audio output directly from coupling capacitor C15.

AUDIO POWER AMPLIFIER

The output of V3A or the audio filter is coupled into the audio power amplifier (*Fig. 2*) through selector switch SW1. I decided to add an auxiliary input to the power amplifier so that I could plug in my FM tuner or CD player (or any line level audio signal). The power amplifier is a simple, classic design that many readers will have seen before. Tube V3B is an input amplifier, and V5 is a "long-tailed" phase splitter. A pair of push-pull 6V6s produces about 9W of audio power.

One slightly unusual feature is that the 6V6s have separate cathode bias re-

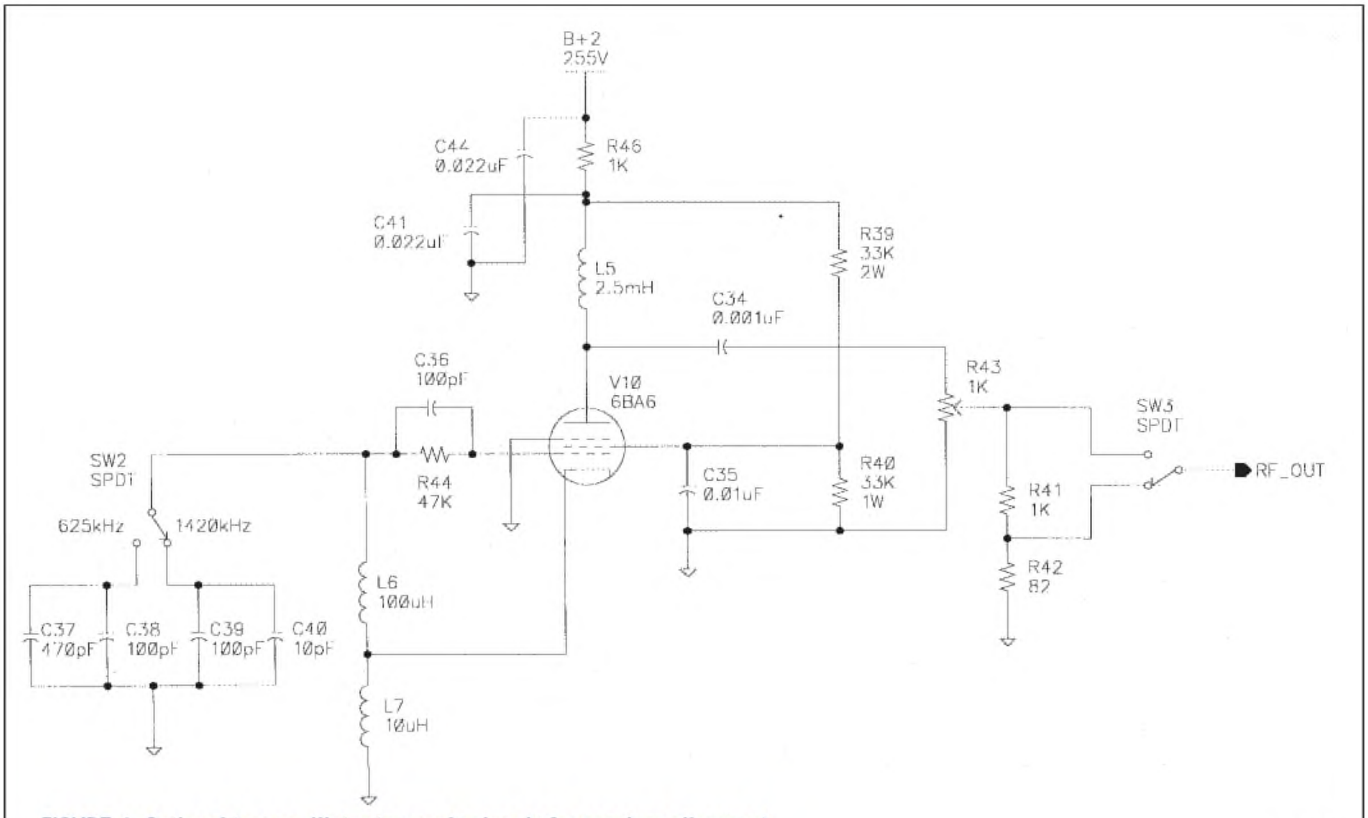


FIGURE 4: Optional test oscillator to supply signals for receiver alignment.

G-2109-4

sistors. I did this to accommodate mismatched tubes. With a single cathode resistor, one tube can hog most of the bias current if the 6V6s are badly mismatched. With separate resistors, you can plug in any two working 6V6s and achieve satisfactory performance.

A 10W Hammond output transformer (type 1608) couples the push-pull 6V6s to the speaker. About 15dB of negative feedback is applied from the speaker output to the cathode circuit of V3B. Note that the output transformer T2 is shown wired for an 8Ω speaker. If you opt to wire it for 4Ω, then R31 should be about 750Ω instead of 1000Ω. If you wire it for a 16Ω speaker, then R31 should be about 1.5kΩ.

POWER SUPPLY

The power supply is shown in *Fig. 3.* A conventional full-wave rectifier and pi-section filter supplies 325V for the power amplifier stages (B+1). Gaseous regulator tubes V8 and V9 produce a shunt-regulated 255V supply for the RF stages (B+2).

I regulated this supply to provide low AC ripple as well as a stable operating point for tubes V1 and V2, so that line voltage and temperature fluctuations wouldn't detune the radio. A regulated supply is probably not essential, however, as long as B+2 is in the 240–270V range. An RC decoupling filter produces 240V for the low-level audio amplifier stages (B+3).

The 6.3V secondary on T3 supplies heater power for tubes V4–V7. Note that one side of the 6.3V winding is grounded. However, you should run twisted pairs for the AC heater wiring instead of running a single wire and grounding one side of each tube's heater at the socket. Running twisted pairs prevents AC heater current from flowing through the ground connection, thereby eliminating a source of 60Hz hum.

TABLE 1
COMPOSITE PARTS LIST FOR *FIGS. 1–4*

C1	Dual ganged 365pF air variable tuning capacitor (RF Parts Co. Part #10295 or Antique Electronic Supply Part #C-V365-X3 or equivalent)
C2	33pF air variable tuning capacitor (RF Parts Co. Part #074-033 or equivalent)
C3, C4, C7, C8, C35	0.01μF, 500V, disc ceramic
C5	15pF, 500V, dipped mica, 5%
C6, C9, C23, C41, C44	0.022μF, 500V, disc ceramic
C10	33pF, 100V, dipped mica, 5%
C11, C36, C38, C39, C43	100pF, 100V, dipped mica, 5%
C12	0.047μF, 100V, film
C13, C16	220μF, 16V, electrolytic
C14	0.01μF, 50V, film
C15	0.22μF, 400V, film
C17–C19	0.1μF, 400V, film
C20, C21	470μF, 35V, electrolytic
C22	4700pF, 100V, film or dipped mica
C24, C26	47μF, 450V, electrolytic
C25	100μF, 450V, electrolytic
C27–C30	100μF, 25V, electrolytic
C31, C32	4700μF, 16V, electrolytic
C33	0.1μF, 100V, film
C34	0.001μF, 500V, disc ceramic
C37	470pF, 100V, dipped mica, 5%
C40	10pF, 100V, dipped mica, 5%
C42	0.001μF, 100V, film
D1	Germanium point-contact diode, 1N34A, or 1N60A, or equivalent. Available from Antique Electronic Supply, RF Parts Co., or All Electronics.
D2, D3	1N4007, 1000V, 1A rectifier diode
D4, D5	1N5401, 100V, 3A rectifier diode
D6–D8	1N4001, 50V, 1A rectifier diode
D9, D10	1N4735A, 6.2V, 1W zener diode
IC1	LF351N, JFET input op amp
IC2	LM317T, adjustable voltage regulator (mount this part on a heatsink or on the chassis, using a thermally conductive, electrically insulating gasket)
L1	Nine turns of AWG22 enameled magnet wire, spaced ⅛″ (3.18mm) apart, wound on the outside of a 24″ × 24″ (61cm × 61cm) square wooden frame. See text.
L2, L3, L5	2.5mH RF choke, Hammond part #1535B or equivalent.
L4	Filter choke, 12H @ 100mA, Hammond part #193B or equivalent.
L6	100μH molded RF inductor, 10%, Dale Vishay type IM2 or equivalent. Mouser part #70-IM2-100
L7	10μH molded RF inductor, 10%, Dale-Vishay type IM2 or equivalent. Mouser part #70-IM2-10
L8	Filter choke, 2H @ 100mA, Hammond part #154M. (Be careful about using a substitute for this choke, since other chokes may have different audio characteristics. See text.)

Q1	2N2907A or equivalent, PNP silicon bipolar transistor
R1	180Ω, ½W, 5%
R2, R39	33kΩ, 2W, 5%
R3, R9, R12, R20, R31, R41, R46	1kΩ, ½W, 5%
R4	220Ω, ½W, 5%
R5, R10, R15, R24, R25	1MΩ, ½W, 5%
R6	470Ω, ½W, 5%
R7, R8	10kΩ, ½W, 5%
R11	33kΩ, ½W, 5%
R13, R27, R28	220kΩ, ½W, 5%
R14	10MΩ, ½W, 5%
R16	4.7kΩ, ½W, 5%
R17	22kΩ, ½W, 5%
R18	100kΩ audio taper potentiometer
R19, R35	100Ω, ½W, 5%
R21, R22	51kΩ, 1W, 5%
R23	62kΩ, 1W, 5%
R26	22kΩ, 2W, 5%
R29, R30	750Ω, 3W, 5%
R32	1.6kΩ, 10W, 5%, wirewound
R33	2.2kΩ, ½W, 5%
R34	270Ω, ½W, 5%
R36	330Ω, ½W, 5%
R37	250Ω trimmer
R38, R44	47kΩ, ½W, 5%
R40	33kΩ, 1W, 5%
R42	82Ω, ½W, 5%
R43	1kΩ linear taper potentiometer
R45	470kΩ, ½W, 5%
SW1–SW3	Single-pole double-throw toggle switch
SW4	Double-pole double-throw toggle switch
T1	540-1600kHz slug-tuned RF transformer (Antique Electronic Supply part #PC-70-RF)
T2	10W audio output transformer, 8000Ω plate-to-plate primary with ultralinear taps, 4–16Ω secondary (Hammond part #1608 or equivalent.)
T3	Power transformer, 250-0-250V @ 100mA, 6.3V @ 3A, 5V @ 2A (Hammond part #270CBX or equivalent.)
V1, V10	6BA6 or 5749/6BA6W
V2	6C4 or 6100/6C4WA
V3, V5	12AU7A or 5814A
V4	6E5 tuning indicator tube (or substitute an analog meter movement)
V6, V7	6V6 or 6V6GT or 6V6GTA
V8	0B2 gaseous regulator tube
V9	0A2 gaseous regulator tube

I included a 6.3V DC heater supply for tubes V1–V3 to produce a completely hum-free radio. A voltage doubler produces about 11V DC from the 5V heater winding on T3, and a standard slow turn-on regulator circuit using an LM317T produces a practically ripple-free 6.3V DC. The LM317T needs a heatsink, which I provided by bolting the tab to a piece of aluminum angle stock, which, in turn, is bolted to the chassis. A thermally conductive washer provides electrical isolation.

Op-amp IC1 requires positive and negative supply rails. The positive rail is from the 6.3V DC heater supply, and a −12.4V DC supply rail is provided by a half-wave voltage doubler using one side of the 6.3V AC heater winding. The voltage doubler supplies approximately −16V, which is regulated down to −12.4V using a couple of 6.2V zener diodes.

Power transformer T3 is heavily loaded, although not overloaded. After prolonged operation, the surface temperature of T3 reaches 50–55°C. Most power transformers are rated for a 60°C temperature rise from ambient, so the temperature of T3 is not a problem, but you should allow for good ventilation as shown in the open design of *Photos 1–3*.

CONSTRUCTION

I began construction of my radio with antenna L1. I built a 24″ × 24″ wooden frame using ½″ × 1⅝″ clear pine trim stock (*Photos 1–3*). For the upright supports I used ½″ × ½″ clear pine trim stock cut to 36″ lengths. You should assemble the frame without metal hardware. I used small nails to hold the frame together while the glue dried, and then I removed the nails.

I maintained the ⅝″ center-to-center spacing of the turns by cutting slots with a hacksaw in the four outside corners of the frame, starting ¼″ from the edge of the wood. This leaves room for ten slots in the 1⅝″ wide trim stock. Prior to winding the antenna, I stained the pine frame, then applied a coat of polyurethane.

I wound the nine turns of L1 with as much tension as I could by hand, and when I finished the wire was fairly taut. I ran the wires from the two ends of L1 down opposite sides of one of the upright supports. You should do this or something similar in order to keep the wires separate and minimize their mutual capacitance. Do not twist the wires together. You can hold the turns in place with spots of quick-setting epoxy at several locations along each side.

After the epoxy dried, I gave the entire assembly another coat of polyurethane to hold the turns perma-

TABLE 2
PARTS SUPPLIERS

ANTIQUE ELECTRONIC SUPPLY
(480) 820-5411
www.tubesandmore.com
RF transformer T1, Hammond components, tubes, air variable capacitors

RF PARTS CO.
(800) 737-2787
www.rfparts.com
Air variable capacitors, 6E5 tube, germanium diode

MOUSER ELECTRONICS
(800) 346-6873
www.mouser.com
Inductors, capacitors, semiconductors, metal chassis

ALL ELECTRONICS
(800) 826-5432
www.allelectronics.com
Germanium diode, various surplus items

JAMECO ELECTRONICS
(800) 831-4242
www.jameco.com
Clarostat conductive plastic potentiometer, capacitors, resistors, semiconductors

nently in place. You should wait, however, until you have aligned and tested your radio before using the epoxy or polyurethane, in case you need to change the number of turns on L1. You can hold the turns temporarily in place by using tape.

Loop antennas are directional, and maximum signal pickup is obtained when the signal arrives in the plane of the antenna. Conversely, there is a null in the signal pickup when the signal direction is perpendicular to the plane of the antenna. I bolted the antenna upright supports to the radio chassis in my receiver, so I need to rotate the entire chassis for best reception. But you may choose to build your radio so that the antenna itself rotates.

Assemble the radio itself on a 12″ × 8″ × 2″ aluminum chassis. You should definitely use a metal chassis to obtain a low impedance RF ground. From *Photo 1*, you can see the RF circuitry is on the left-hand side of the chassis, with tubes V1 and V2 at the left rear.

The main tuning capacitor C1 mounts on top of the chassis inside a 3″ × 4″ × 5″ aluminum box. The box provides a ground shield around C1, protects it mechanically, and keeps dust from accumulating between the plates. The power-supply transformer T3 and choke L4 are at the right front of the chassis, and output transformer T2 is at the right rear.

Proper grounding is essential in any RF or audio circuit, and having both on the same chassis as the power supply complicates matters. You should keep all RF grounds short (less than ½″) and make them directly to a solder/ground lug bolted to the chassis. Make grounds for audio tube V3 in the same way.

Isolate the ground for the auxiliary audio input jack from the chassis with a nylon washer and run a wire to the same ground lug that R18 and R19 connect to. You should make all grounds for the power supply and audio power amplifier (tubes V5–V7) to an AWG 14 (or larger) copper bus wire, which connects to the chassis at a single point, preferably at the same ground lug that R18 and R19 connect to. Connect the ground wire in the AC power cord directly to a ground lug.

For best operation of the radio (as well as for safety reasons), the radio needs a grounded three-wire line cord. If you live in an older house and your electrical system is ungrounded, you'll need to provide an alternate ground for the chassis. The *Radio Amateur's Handbook* describes safe and effective ways of doing this.[5]

Keep wiring for the RF circuitry as short as possible—ideally ½″ or less—but I have found that it was not always practical to do this. *Photo 4* shows an underside view of the chassis, with my RF wiring in the right rear. The wires leading to capacitors C1 and C2 and antenna L1 are the longest ones. I kept these wires away from the chassis and away from other wires and components in order to minimize stray capacitance.

SETUP AND ALIGNMENT

To set up and align your receiver, you'll need a high input impedance (10MΩ) vacuum tube voltmeter (VTVM) or digital multimeter (DMM). You'll also need an RF signal generator to provide test signals at the low and high ends of the AM band (approximately 600 and 1400kHz). An oscilloscope is not necessary, but one can be very helpful in troubleshooting.

If you don't have an RF signal generator, you can build the circuit of *Fig. 4* to supply the test signals. This circuit is a Hartley oscillator built with a spare 6BA6 tube and using the radio's own B+ and heater supplies (remove one of the radio's 6V6 tubes while you power this oscillator from the radio's supply rails, to avoid overloading the power supply).

To prevent direct pickup of the oscillator signal, you should provide shielding by building the oscillator in its own separate metal minibox. If you build the oscillator with close tolerance components as specified, the test signals should be within 7.5% of 625kHz and 1420kHz. If you have an accurately calibrated oscilloscope, you can check the output frequencies to be sure they're correct. If not, you must trust that the oscillator is within tolerance.

When you first power-on the receiver, check the supply voltages to be sure they are correct (±10%) as shown in *Fig. 3*. Adjust trimmer R37 to set the DC heater supply to 6.3V. Also check the voltages at selected points in *Figs. 1* and *2*, which should be within about 10% of the values indicated.

Specifically, the 6V6 cathodes should be at 22V, the V5 cathode at 92V, the V3B cathode at 3.3V, the V3A cathode should be at 4.0V, the V2 cathode at 59V, and the V1 cathode at 2.6V. If these voltages are all approximately correct, all of the tubes in the receiver are probably wired properly. If not, look for a wiring error associated with the respective tube.

To align the receiver, connect either your RF signal generator or the circuit of *Fig. 4* to the ungrounded side of antenna L1 through a 100kΩ resistor (put the resistor on the receiver side of the wire or cable you use). If your tuning capacitor C1 has trimmer capacitors, adjust the trimmers for minimum capacity (fully out). Attach your VTVM to point "A" on *Fig. 1*, and set C2 to its half-meshed position. Apply a ~600kHz signal and tune C1 for the maximum negative voltage, which corresponds to maximum eye closure on V4.

You may need to turn down the signal generator level to avoid overlapping the lighted portions of the eye. Conversely, you may need to turn up the signal generator to see any indication on V4. When correctly tuned, C1 should be almost completely meshed. Be careful to avoid tuning to a second harmonic of the signal generator at 1200–1250kHz.

Using a hexagonal adjustment tool (available from Radio Shack as part of a color TV alignment tool kit), adjust the slug on T1 for maximum eye closure. Then readjust C1 for maximum eye closure, then adjust C2, then T1. Continue adjusting these three in sequence until the eye closes no further. You may need to turn down the signal generator as you approach alignment. Temporarily mark the position of C2.

Apply a ~1400kHz signal and adjust C1 for maximum eye closure, then adjust C2. Alternately adjust C1 and C2 until the eye closes no further, and temporarily mark the position of C2. Make sure that maximum eye closure is obtained before C2 is fully meshed. If maximum eye closure is obtained only with C2 fully meshed, you'll need to add some additional capacitance in parallel with C2.

Try adding a 10pF dipped mica in parallel, then repeat the alignment procedure. But I doubt that you'll need any additional capacitance if you used a 33pF trimmer for C2. Assuming C2 is not fully meshed, alignment is complete. In my radio, C2 is at about one-third meshed at the low end of the AM band and at two-thirds meshed at the high end.

OPERATION

At this point you should be able to receive many AM radio stations. Tune the radio using C1, then adjust C2 as a fine-tuning control. A little practice will make this easy. Using radio stations as markers, you can determine the tuning range of your receiver.

If you find that you cannot tune to the top end of the AM band, you can remove either a full turn or half turn from L1. Conversely, if you cannot tune to the low end of the AM band, try adding a full or half turn to L1. You'll need to realign the receiver if you change the number of turns on L1. As I mentioned earlier, tapping L1 is one method to extend the tuning range of the receiver, but the need to realign when the taps are changed limits the practicality of this approach.

Once you've settled on the number of turns for L1 and made your final alignment of the receiver, you can make a calibrated dial for C1, and perhaps for C2 as well. I simply affixed a piece of paper to the aluminum chassis behind the knob for C1 and marked my favorite stations on it. But you could easily come up with a more professional-looking dial.

You'll find that the receiver has more gain and greater selectivity at the low end of the AM band than at the high end, but this does not seem to be a serious problem in practice. The reason for this variation is found in transformer T1. Since the primary of this transformer needs to be resonated at a frequency below the AM band, the effective load on V1 varies as a function of frequency, and hence the gain varies.

It is also possible to wind a transformer with a primary that is resonated above the AM band. In fact, that was the more common approach back in the early days of radio. But that only reverses the problem, moving the high gain end of the band to the top.

This variation in gain and selectivity due to the coupling transformers is one of the limitations of a TRF receiver. I find the gain and selectivity more than adequate across the entire band. But if your favorite station is weak and near the top end of the band, you may find the gain or selectivity inadequate. I think that is unlikely, but you'll only find out for certain by building a prototype.

As I mentioned, the strongest RF signals in my location produce about −1.5V at point "A" in *Fig. 1*, and I set the gain of IC1 to produce full eye closure for this signal level. You may find that you have stronger or weaker signals, and you can adjust the value of R17 to increase or decrease the gain of IC1 and keep the tuning eye in a useful range. If you live in an area with a very strong AM station, you may get more than −3V at point "A," which is about the largest signal that the audio amplifier V3A can handle without distortion.

If that is the case, you have two options for handling the large signals. One option is to use a 1MΩ potentiometer in place of R10 to allow you to attenuate the audio signal (connect the wiper to the grid of V3A). The second option is to add an RF gain control by placing a variable resistor in series with R1. Connect one end of a 5kΩ potentiometer to ground, and connect both the wiper and the other end of the potentiometer to the previously grounded end of R1. When the potentiometer is set for zero resistance, the circuit will work as it does now. As the resistance is increased, the gain of V1 is reduced.

If you do add this RF gain control, you should use a high-quality potentiometer, because DC current flows through it at all times, and standard carbon pots quickly become noisy under such conditions. You'll also want to make sure that the potentiometer has almost exactly zero resistance when it is set to one end of its rotation. I recommend a Clarostat or similar conductive plastic potentiometer. Avoid wirewound potentiometers because they are inductive. You can also use a switch to select resistors of different values for R1.

AUDIO QUALITY

To get the best audio performance from this receiver, I recommend a high-quality speaker, although it need not be expensive. Most AM radios have poor speakers. I use a Madisound "Sledgeling," a small two-way bookshelf speaker. The audio quality you get will also depend on the RF signal strength and how much RF interference there is in your location, and both of these factors can vary from day to day.

Finally, the audio quality you get depends on the quality of the signal that the radio station itself transmits. Almost all radio stations (AM and FM) use some sort of signal processing to increase the average modulation percentage of their signal. This puts more power into the sidebands and increases the effective amount of power that the station is transmitting, making the station sound louder by increasing the signal-to-noise ratio.

There is nothing really wrong with this, but it obviously impacts audio quality, and if it is overdone the quality can be poor. Stations also sometimes indulge in equalization, effectively turning up the bass and treble controls whether you like it or not. I find that WNYC and other NPR stations broadcast high-quality signals.

CONCLUSIONS

There are several options for building your radio. Obviously, you can build the entire receiver as shown in *Figs. 1–3*, as I did. You could also use the AM tuner as part of your audio system and omit the power amplifier circuitry in *Fig. 2*. In this case the power transformer T2 and filter choke L4 can both be smaller (a 50mA current rating would be sufficient). If you just want to experiment, you could build the tuner circuitry in *Fig. 1* and power it from one or more bench power supplies, but you should still use a metal chassis.

An AM receiver may seem like an anachronism in these days of radio programs streamed over the Internet, but it has brought pleasure to me and to my wife. Perhaps the apparent anachronism is part of the pleasure. In spite of the radio's simplicity, it is the most sensitive radio in the house. The large antenna and a sensitive detector are the keys to its RF performance.

On recent winter evenings, my wife and I have listened to a French language station from Montreal, which is over 300 miles away. It is certainly possible to build a more sensitive receiver using the superheterodyne approach, but I doubt you'll be able to beat the audio quality of this receiver. It is not CD quality, of course, nor even FM quality, but I think I can promise something much better than you've come to expect from AM radio. I think you will be most pleased with it. ❖

REFERENCES

1. Vince Roberts, "Letters," *audioXpress*, February 2001, p. 81.

2. My favorite regenerative receiver appeared on p. 210 of the 1944 *Radio Amateur's Handbook*, although this reference is now practically unobtainable.

3. The invention of the superheterodyne was the work of Edwin Howard Armstrong, who also invented FM. See *Man of High Fidelity* by Lawrence Lessing, Bantam Books, Chapter 7, 1969.

4. *The ARRL Handbook for Radio Amateurs*, ISBN #0-87259-186-7, p. 17.18, 2000.

5. *The ARRL Handbook for Radio Amateurs*, ISBN #0-87259-186-7, p. 9.3, 2000.

Extending the 30″ Tractrix Horn

This author shows you how to lengthen his horn design to increase its compatibility with other drivers.

By Robert Roggeveen

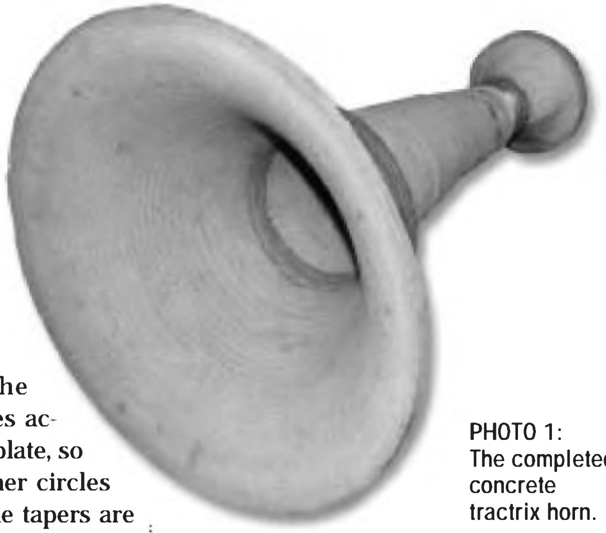


PHOTO 1: The completed concrete tractrix horn.

A large mouth concrete axisymmetrical tractrix horn with a small throat is likely made up of two sections: the neck and the bell (*Photo 1*). The reason is twofold: weight and manufacture technique as a result of gravity. The neck is the length from the throat to roughly halfway along the axis toward the mouth. The bell is the rest of the expansion to the terminus mouth flair.

This article describes how to make an extension to the tractrix horn described in "Build Your Own Axisymmetric Tractrix Horn" (*aX* Sept. '02, p. 28); in particular, to the 30″ diameter mouth with 9⅝″ throat (*Photo 2*) that was mated 1:1 to a 12″ EV. (I misstated the diameter as 36″ in the article. It really measures 30″. Sorry for the mistake.)

STRETCHING THE NECK

Applying theory, you determine that the lower cutoff of a 15″ radius mouth is 150Hz. By lengthening the neck along the tractrix curve path and thus effectively narrowing the throat's diameter, you can mate this large horn to various other drivers needing a smaller throat. I chose a 2⅞″ diameter throat.

This neck section has two wooden plates—one for the driver and one to connect with the bell section. Use ¾″ marine plywood for the plates. In between the two plates is concrete, molded to follow the tractrix curve. The mold is built up of sections of disks, each with a radius of 15″ made into several cones and stacked on top of one another. Because of the nearly vertical shape, you also need a retention mold to keep the mortar from falling off, which is what sets this project apart from the first.

You must transpose the bell's eight mounting holes accurately to the connecting plate, so when you fasten it the inner circles match up smoothly and the tapers are facing the same direction. Ideally, these two wooden rings are made by joining two ¾″ plywood disks of 15″ diameter, cutting the inner throat circle and mounting holes for the driver (12″ EV), at once. But I decided on a different process.

MAKING CONNECTION

I used a test bolt, cut short to ¾″ with a dull point ground on one end. Once you have inserted and screwed it into one of the eight T-nuts in the bell's throat plate, with the ground tip reaching slightly above the plate surface plane, lift the connecting plate, align the center holes by lifting and shifting, and then lower it. Once it is aligned, press lightly where the bolt is inserted. This leaves an indent that marks the spot to be drilled out, slightly oversized relative to the ¼″ hex bolt used to connect the two plates.

Remove and insert the test bolt into the T-nut at 125° to the first, again slightly raised above the plane. Insert a hex bolt through the hole drilled in the connecting plate and into the first T-nut. Since that hex bolt allows the connecting plate to move slightly, align the inner circles and press down gently where the test bolt is inserted, leaving an indent which you will again drill out. Repeat this sequence for all T-nuts, each time placing more hex bolts through the added drilled-out holes in the connecting plate.

The connecting plate (*Photo 3*) looks like a ¾″ thick flat round washer with a 15″ outer diameter and 9⅝″ tapered inner diameter. The slant of the taper is about 70° to the surface plane. There are eight 5⁄16″ diameter holes equidistantly placed on a virtual circle corresponding to the T-nuts embedded in the bell's throat plate, roughly in the center of the flat washer's ring area.

This virtual circle produces two areas on the flat connecting plate: an inner surface from the 9⅝″ diameter hole to the virtual circle and an outer surface from the outer circumference to the virtual circle. The inner surface gives the image of what is roughly the

PHOTO 2: Horn extension inner mold and connecting plate.

cross section of the concrete horn wall at the connecting plate, revealing a 1¼″ thickness.

Thirty-two drywall fasteners are screwed equidistant onto this inner surface area of the connecting plate: 1½″, 2″, and 2½″ in length at a 17° angle with the axis of the horn, slightly more steep compared to the 20° inward slant of the inner ring. These are drawn as far into the predrilled holes and placed in a sine rhythm: 2½, 2, 1½, 2, 2½, 2, 1½, 2 sequence. All slant inward with the contour of the horn's neck. If they reach outward into the open, then you've applied them to the wrong side of the wooden ring. Think of all these screws as anchor rods embedded in the concrete, holding the wooden ring to it.

The eight mounting holes in the connecting plate are fitted with the following ensemble: a hex bolt, a washer, two spacers, and a nut (*Photo 4*). This arrangement is necessary because once the mortar is applied enough space needs to be left so that you can turn the hex bolt with a ⅛″ socket. The height of the tube spacer is ±1½″—cut from ½″ inner diameter, clear plastic hose (⅛″ thick) that I had on hand. The other spacer is a cylindrical form to stabilize the tube spacer from the inside. Except for painting the connecting plate with bonding agent, it is ready for assembly.

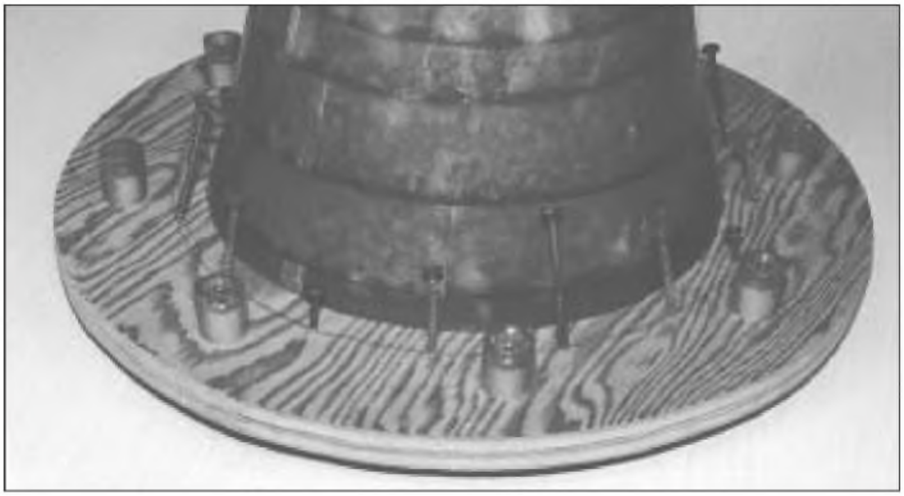


PHOTO 3: Close-up of connecting plate with fasteners.

CONSTRUCTION AND ASSEMBLY

Knowing the area of the inner circle in the connecting plate, you can deduce the circumference. Because the inner circle is cut under a slant, use the larger of the two circumferences. Like the bell, you will make the neck of 30″ diameter paper disks sections. From a 30″ diameter paper disk mark a section with the circumference length slightly longer, by about 1″, than the total length of the inner circle circumference of the connecting plate.

In addition, use a protractor to draw a circle with a 7″ radius in the center of the 30″ disk. For later sections, as they become smaller, use a 9 to 10″ radius. Cut out what looks like a washer section. Mating the linear edges of the section with two paper clips will produce a truncated cone. For a thorough explanation of how to make cones and the tractrix curve for these concrete horns, read the *aX* Sept. '02 article.

Slip the connecting plate over the cone. It should not slide off at the bottom. If it gets hung up with more than ¼″ remaining under the plate, adjust the paper overlap. It helps to mark several radial lines on the paper section to keep the cone properly aligned when you make adjustments. Cut the excess, if any, to leave a ¼″ glue strip.

This is the initial cone. Before gluing it, copy this paper section on the leftover section of the 30″ diameter disk you just worked with. You can cut about four or more cone sections from the same disk. Each cone is ½″ less in circumference than the previous one. So, following the initial cone, the second cone is ½″ less in circumference length, the third is 1″ less than the initial circumference length, the fourth is 1½″ less than the initial circumference length, and so forth.

All cones stacked up produce a shallow axisymmetrical tractrix curve of the neck of the horn along its 24″ length. Prepare the stacked cones with epoxy for a sturdy and water-repellent mold. It is now ready for assembly.

The outer retainer mold has a base cone that fits outside the 32 screws and just inside the eight mounting holes of the connecting plate (*Photo 5*). For the retainer mold a larger diameter paper disk is used to make the cones. They have at this distance along the axis a greater base circumference. I estimated 48″ diameter and worked with sections of that size. The shape of the outer retainer mold is not critical and is built up in four sections (*Photo 6*).

Working with cones of 6¼″ in height allows for adequate though tight space to work mortar into the void formed by the inner tractrix mold and the outer retainer mold. Once the base cone has been established, the second cone will fit over the base cone at its very top (*Photo 7*), producing an overlap of just ¼″. The third cone overlaps the second cone's top ¼″. And the fourth overlaps the third (*Photo 8*).

Whereas building the tractrix curve mold cones are stacked on one another leaving only a small conical area exposed at the bottom of the previously placed cone, the building of the outer mold is one where stacking cones leaves the vast majority exposed, overlapping only ¼″ or small conical area at the top of the previously placed cone.

COVERED WITH MUD

Consider the substantial dry weight of the concrete. The 30″ bell weighs 63 lb. The extension weighs 40 lb. This outer set of cones does not have the integral strength of the inner tractrix mold. Furthermore, the cones of the outer mold are not glued to each other, so you need to reinforce them with lateral rings perpendicular to the cone surface extending outward (*Photo 6*). Cut paper rings and glue them with epoxy onto the outer surface of the individual outer mold cones. Paint the cones on the inside with epoxy to strengthen them and make them water-repellent.

In addition, applying mortar causes the outer cones to move upward relative to the inner mold! To counteract this force, you may choose to apply tabs to the base outer mold, held in place by the eight hex bolt ensembles. I did not have these tabs in place, causing the mold to move ½″ upward when I worked the mortar into the gap.

Take the prepared connection plate with its screws, bolts, and spacers in place, and fit it into position onto the

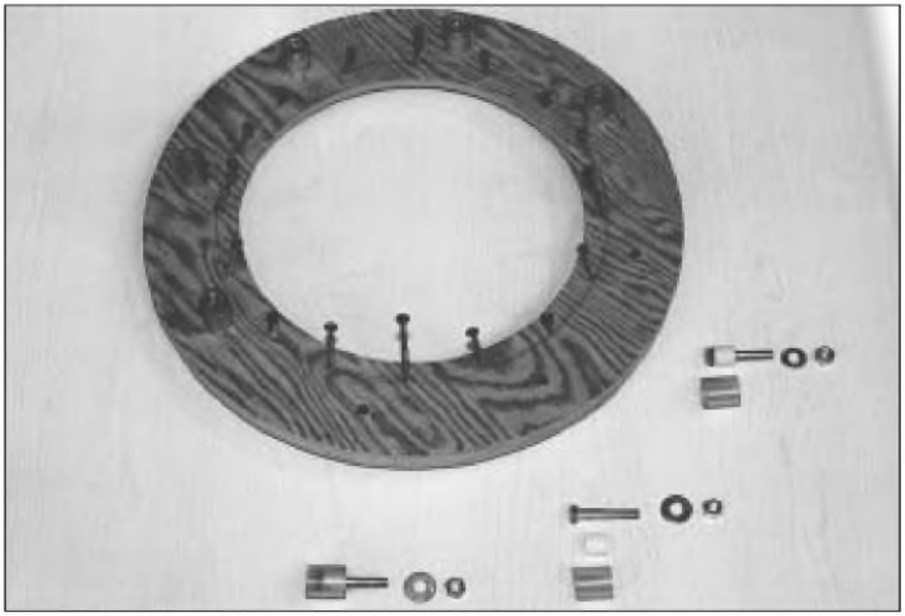


PHOTO 4: Bolt mounting and casting spacer sequence.

PHOTO 5: Fitting the base retainer mold sections, reinforced by a second ring.

epoxied tractrix inner mold, at the base. Just prior to applying mortar, paint the inner surface area and screws on the connecting plate with a bonding agent. Note the outer mold cones with the paper reinforcement rings mounted. These are painted with epoxy on the inside. A ¼″ plywood disk 6″ in diameter with a 2⅛″ circular hole in the middle flattens the mortar level at the throat, once applied.

Start by applying mortar into the narrow upright wall void formed by the outer base cone tractrix and tamping and thoroughly working the mud around the screws at the base and all along the neck upward in a spiral motion. Stack the four outer mold cones sequentially as you build up the mortar to form the outer retainer mold. The greater the height, the more precarious it is to work the mortar. The height of each cone just barely worked. I used a palette knife to shove mud into what is essentially a curved slot, poking and mixing, laying and pressing fairly swiftly, moving around the perimeter. After building up the first 5½″ of horn wall, place the second outer cone, overlapping the first by ¼″.

PHOTO 6: Retainer mold cone sections.

Measuring, marking, cutting, and gluing paper are all quite important, just as with the inner tractrix mold. Good accurate craftsmanship pays off. Once you toss a portion of mud into the wall void, work it right away, keep the layering manageable, and don't forget where you just were. Some of this kind of work is done by feel, as if blindfolded (*Photo 9*).

FINISHING TOUCHES

At the very upper part of the neck, at the throat, use the wooden level ring to produce an even plane. It helps to have marked the inner mold at the 2⅞″ diameter level. I overapplied mortar, compressing the concrete slightly down to the proper mark. Stick six tack nails into the mortar near the throat on the outside perpendicular to the outer mold surface, pricking through the paper cone. Let it dry.

Install the driver mounting plate with T-nuts embedded in the ¾″ marine plywood as needed. My prototype included 12 to test the Siare, Pyle, and JBL drivers. Prepare the plate with anchor screws. Place a paper ribbon around the perimeter of the plate to produce a dam to keep the mortar from spilling off the surface of the plate.

Break the concrete extension out of both molds. Let it dry some more. If there are no voids you did a most excellent job. Even if there are some big neglected spots where the mud was not mixed in well you can fill mortar in later to repair these blemishes (*Photo 10*). It is essential, however, to have the inside surface finished as smooth as possible for the sound wave to propagate properly. The final horn surface is finished with a paste used to seal bathroom tile grout.

Place the throat end of the concrete extension on a piece of cardboard (*Photo 11*). It may be thin walled, so be careful. The best defense against brittle mud is a good mixture and adequate curing time, three to four days. Check how it looks. Does it stay upright or lean? This is why it was important that the throat was tapped level.

BONDING THE DRIVER PLATE

With the hole centered, T-nuts embedded, wax plugs inserted, and screws and paper edge sticking up, place the concrete throat onto the wooden throat. Mix mortar and paint the plate side facing the concrete and the concrete on the outside of the horn about 4″ up with bonding agent. Place the horn, accurately aligning the throat opening.

Apply the mix in manageable steps, working it around the screws using up and down movements with a palette knife, a dowel, or a piece of wood. Tamp and stir, applying the mix around and up the neck. Before the mortar hardens, you can finish off the top surface in a number of different ways: leave it rough, smooth it, or cut patterns or letters into it. Let it dry. After a day remove the paper dam.

PHOTO 7: Assembling the horn mold pieces ready for casting, a dry run.

PHOTO 8: Overlapping cones in place, a dry run.

PHOTO 9: Forming the concrete mold.

When the concrete has cured and it appears solid, fill in gross, moderate, and minor blemishes as needed on the inside and on the outside of the horn. Fix the substantial blemishes with mortar mix. Let it dry and finally cover all of the horn inside with a finishing patch material. A smooth surface, with no air leaks, is important. Use a narrow palette knife to strike off

PHOTO 10: Patch up any blemishes on your concrete shell.

PHOTO 11: Aligning the wooden and concrete throats.

any excess humps or bumps. Sand the area.

CONCLUSION

This extension project pushed the limits a bit. All mold materials, although fortified with epoxy, are made of paper, so weight plays a major part. The mortar casting is most exciting and anxiety provoking, as about 40 lb of mortar is controlled in paper. To see the paper stand up to such tensions is quite amazing.

I used two layers of cotton duck material, such as from old jeans, to make a gasket to fit between the neck and the bell. Stitch the two at the edges and make eyelets for the bolts. The inside diameter should be flush with the wooden rings. Connect the extension to the bell with eight hex bolts.

Now this is a formidable tractrix horn approximation. It has a 2⅛″ throat, 30″ mouth, is 37″ in length, and weighs 100 lb. The weaknesses are that the horn is slightly foreshortened relative to the ideal, and it is heavy. Strengths are firmness, high decibels in the bandpass, on-axis circular propagation of the sound-wave front, sound presence, and punch at a relatively far distance.

Table 1 contains 11 response tests I conducted on six different configurations so far. I performed the tests in an open field with salal, huckleberry, cedar, and fir at the perimeter 70′ or

TABLE 1

TEST: HERTZ	1 dB	2 dB	3 dB	4 dB	5 dB	6 dB	7 dB	8 dB	9 dB	10 dB	11 dB
61			50	66	56	54		56	62	67	
127	65	76	55	75	79	74	54	78	78	79	
251	73	84	67	80	89	86	71	91	90	92	92
499	92	85	81	94	95	93	82	95	95	91	92
997	92	93	85	88	93	92	86	93	88	90	103
1999	89	90	84	69	88	83	83	85	76	88	97
4001	83	86	86	81	86	80	84	84	75	76	92
7993	76	66	85	64	72	73	83	80	64	67	73
10007	80		76		77	78	66	73			70
12503	56		61		54	63	65	58			
16001			61			60	64				
Throat (in. dia.)	2.12	2.12	4.25	2.12	2.12	2.12	2.12	2.12	2.12	9.37	1.5
Mouth (in. dia.)	12	9.25	18	30	30	30	30	30	30	30	30
Length (inches)	8.75	26	10	37	37	37	37	37	37	11	45
Gap (inches)	—	—	½	—	—	—	½	—	—	—	—
Back vol (quarts)	—	—	—	—	—	—	—	2	2	—	—
f_C(Hz)	358	458	238	150	150	150	150	150	150	150	150

Drivers:	Test:
JBL 104H-2	1, 2, 5, 8
SIARE 16-VR	3, 6, 7, 11
PYLE W6C200F	4, 9
EV 12L	10

more away from all sides. It was a sunny, fall afternoon (*Photo 12*).

Although designed as a tractrix contour propagation, the extension mimics an exponential expansion fairly closely. So I tested it as such (test #2): 2⅛″ throat, 9¼″ mouth, and 26″ long!

For the test I used a CBS test CD (Old Colony) and played on an unmodified CDB 560 Magnavox CD player. The tube amp is a 145 Leslie, taken from a tone cabinet. I measured with a Radio Shack digital sound level meter (33-2055) on-axis at 1m. Signal potential is held at 2.71V.

This extension project spurred me on to experiment further. Test 11 is of a horn incorporating an additional 6″ extension narrowing the throat from 2.12 to 1.5″ in diameter. I may report on this later. In the meantime, enjoy building and listening to horns. ❖

PHOTO 12: The author tests his finished product.

Improving Dynaco's FM-5 and AF-6 Tuners

Discover how to add IC voltage regulators to either a Dynaco model FM-5 or model AF-6 tuner, and also correct the low-frequency drop in the demodulated signal of the early model FM-5. **By Daniel Dufresne**

A few years ago a friend gave me a Dynaco FM-5 FM tuner, which I cleaned up and tried. The unit worked and provided mono audio output, but no stereo output. I replaced the stereo decoder IC Q205, a Motorola MC1307P, with an NTE Electronics Inc. NTE722, and I re-aligned it. It worked fine and I installed it in my dining room, along with a Dynaco SCA-80 integrated amplifier and two Dynaco A-25 speakers, a true classic all-Dynaco system. I know this sound system is more than 30 years old in design and production, but it works and has escaped the landfill.

Later, I noticed that if I turned on the tuner and immediately changed station tuning, after about five or so minutes it would start muting occasionally. After ten more minutes, it would mute permanently. All I needed to do was to retune slightly and it would stay properly tuned and not mute until I turned it off.

I told myself that I probably had been inattentive when I initially aligned it and I needed to realign it. Then I thought that no misalignment would cause this, but drift would show these types of symptoms. Could the power-supply voltage drift and cause the rest to drift, too? I do not have the equipment to check the RF or IF circuits for drift, but I sure can check the power supply.

VERSIONS

My FM-5 is the early model with PC-20 and PC-21 printed wire boards (PWBs). There is a later FM-5 version with a different set of boards, PC-25 and PC-26, which are the same ones used in the AF-6. The information in this article can be used for all versions. The AF-6 has a different front end and an added assembly, PC-27, to take care of the AM tuning and demodulating functions.

REFERENCED DOCUMENTS

I checked back issues of *The Audio Amateur* for any information on Dynaco tuners or FM tuners in general. "Upgrade your FM-5"[1] is a two-page article on improving sonic qualities. There is a "Kit Report"[2] on the VSM Audio, MPX-2 phase-locked loop stereo demodulator using an KB4437 IC, designed for replacing the stereo decoder circuit with an improved IC-based stereo decoder. This was followed by a letter[3] that describes experiments done on this kit and other notes on stereo adjustment for the FM-5.

Finally, I found an article, "Upgrading your FM tuner,"[4] on general modifications. It shows how to add signal buffers on audio outputs and certain internal nodes where there are some notable improvements. There is a section on adding a regulated power supply, tuning indicator, and a muting comparator. The regulator is based on the Raytheon RC4195-TK IC, which is a fixed dual 15V positive and negative circuit. Raytheon sold the rights for this IC to Fairchild Semiconductor, but I could not find it on their website.

The Motorola MC-1468 and the National Semiconductor LM325 are equivalents. The LM325 will be obsolete shortly. I could not find the MC1468 on the On Semiconductor website.

I found nothing on drift or erratic muting behavior either in the literature or on the web. I did find the schematic for the FM-5 with PC-25 and PC-26 and tuner alignment information at these addresses: http//home.indy.net/_gregdunn/dynaco/components/FM5/align.html, and http//members.home.net/dunn.greg/fm5/schem.jpg. (Hint: go to these addresses and save both the tuning procedure and the schematics on your computer for safekeeping, in case this site disappears.)

JUST MAYBE

Having put forth the hypothesis that supply drift was the cause of the misbehavior, I needed to investigate. This means some measurements and some experiments. I also needed an explanation for drift. Even if the FM-5 is an all-

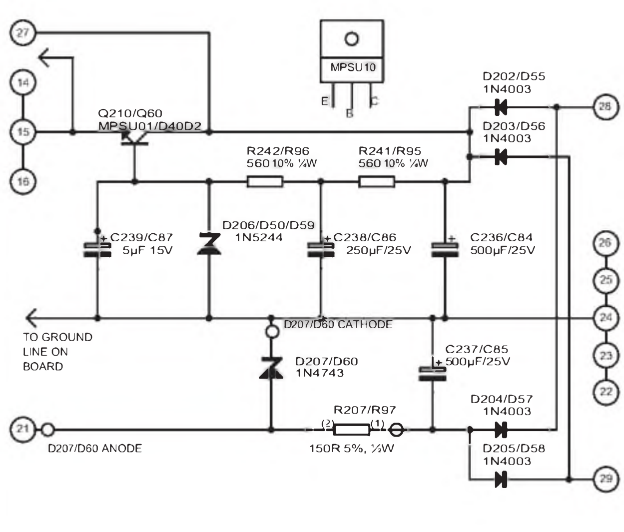


FIGURE 1: Part of original circuit. A-2137-1

semiconductor tuner, thermal drift is everywhere—in resistors, diodes, and transistors alike. Check some data books, and you will see. You need to determine and evaluate how big and how important its impact may be on circuit performance.

I looked at the tuner's schematic and noticed that it uses a simple shunt zener diode for the negative supply and an amplified zener for the positive supply (*Fig. 1*). The parts identifications refer to FM-5 both PC-21 and PC-26 in that order, except for D206/D50/D59, which is for FM-5 PC-21/FM-5 PC-26/AF-6 PC-26. The numbered circles represent the PWB eyelets used for interconnections. I brought the tuner to my test bench and did some measurements. The results are in *Tables 1–5*.

ORIGINAL CIRCUIT DESCRIPTIONS (PART REFERENCES ARE FOR FM-5 PC-21 VERSION.)

All the power for this tuner comes from a single power transformer with dual secondary. One winding powers the panel lamps; the other has a center tap and powers all the electronics. A full-wave rectifier bridge made from four discrete diodes, D202–D205, charges up two capacitors, C236 and C237, to provide the two raw DC voltages, positive and negative. A two-stage low-pass filter, made from R241 and C238 and R242 and C239, respectively, powers zener diode D206, which connects to the pass transistor's, Q210, base. The collector connects to the raw positive supply. The emitter is the output terminal.

The negative supply is made of a series-limiting resistor, R207, and a shunt zener diode, D207. These two circuits provide a positive supply of 13.6V and a negative supply of –13.6V.

POWER-SUPPLY MEASUREMENTS ON ORIGINAL CIRCUIT

The raw positive supply is typically 18.8V with about 1.2V peak-to-peak ripple. The raw negative supply is 22.6V with 250mV peak-to-peak ripple. All these values can vary according to the supply line voltage value.

The positive supply loads are: RF front end only, 16.7mA–18.3mA; PC-20 and front end, 63.7mA–65.6mA; and total positive supply load between 93.7mA and 97.4mA. The two lamps, "tuned" and "stereo," are powered from the raw DC and consume about 36mA each. The negative supply load is 9.6mA–10.2mA (into PC-20) and 9.7mA–10.4mA (total).

I measured the warm-up drift, which caused the positive supply to change by 0.3V after three hours. Power-line regulation showed a 0.3V change for a 105V to 130V input change range. I told myself that for the tuner's cost and year of design, it is reasonable to use simple zener voltage regulator circuits.

Integrated regulators cost very little these days and could improve the per-

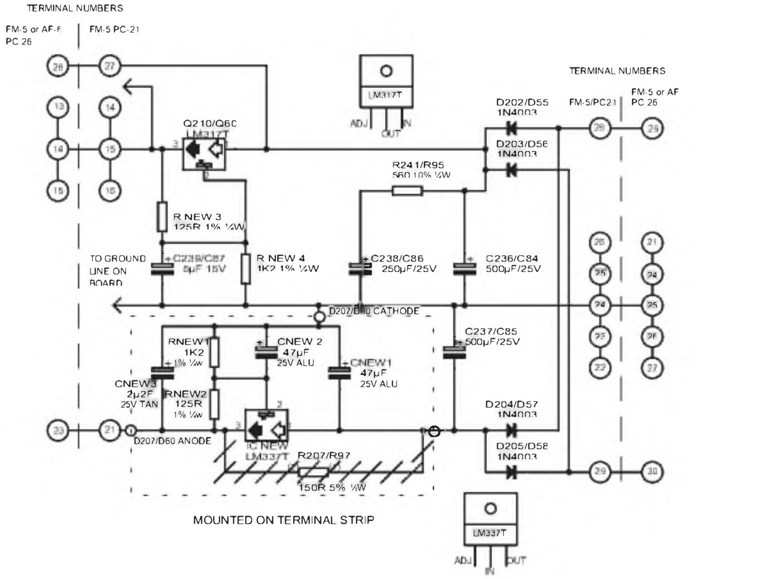


FIGURE 2: Modified circuit.

A-2137-2

formance of this tuner's power supply. I decided to replace both the positive and the negative regulator circuits with IC regulators. *Figure 2* shows the schematic of the new voltage regulator section. Whether this change would eliminate the observed drift and improve measured or audible sonic performance remained to be seen.

Should your tuner have a power-supply failure, you might also consider these modifications, which add little cost and take two to three hours to execute. The modified circuit performance along with the original circuit measurements are in *Tables 1–5.*

NEW REGULATOR CIRCUIT DESCRIPTION

The positive voltage regulator IC is an LM317T, which is adjustable simply by selecting the appropriate resistor values. The first resistor is connected between the regulator output and the regulator adjust terminal. I chose a 124Ω value because the regulator reference voltage is nominally 1.25V. This choice makes the rest of the calculations quite simple.

Also, the regulator needs a minimum of about 10mA to work properly, and this low resistor value provides such a load. To get 13.2V you need the second resistor connected between the adjust terminal and ground to be 1k2. It will have approximately 12V across its terminals. The adjust pin is bypassed to ground with a 5μF capacitor (C239 or C87) to improve the ripple rejection.

The negative voltage regulator circuit is the same as for the positive, but with two more capacitors: one on the input and one on the output. Both capacitors improve and ensure regulator stability because of the added wire resistance and inductance.

CAUTION! GLASS INSIDE

When working inside the tuner, try to stay away from the two tubular dial lamps, Dynaco part number 526008, which are located at both ends of the tuning scale. The bulbs look like longish fuses, 1.625″ long by 0.25″ in diameter. The lamp bulbs are somewhat fragile; they break easily at the glass to terminal junction. The terminals break away from the glass body and you are left with a short piece of wire protruding from the body; that is, if you are lucky. If not, there is no wire left attached to the body for you to try to reattach.

New bulbs are difficult to find. If you know where to find replacement lamps for these, please contact *audioXpress* with the manufacturer's name, address, part number, price, and distributors.

TABLE 1
POSITIVE SUPPLY, DRIFT MEASUREMENTS

	ORIGINAL CIRCUIT VOLTS, DC	LM317 VOLTS, DC
Initial	12.88	13.2023
After 3 hours	13.18	13.2112
Total drift	0.3	0.0089

TABLE 2
POSITIVE SUPPLY, LINE REGULATION MEASUREMENTS

LINE VOLTAGE VOLTS, AC	ORIGINAL CIRCUIT VOLTS, DC	LM317 VOLTS, DC
105	12.80	13.1959
120	13.09	13.2009
130	13.12	13.2059
Regulation	0.32	0.010

TABLE 3
POSITIVE SUPPLY, HUM AND NOISE MEASUREMENTS

	ORIGINAL CIRCUIT	LM317	IMPROVEMENT
Measurement bandwidth	mV, AC	mV, AC	dB
22.4Hz to 22.4kHz	0.85	0.049	24.7
Plus "A" weighting	0.64	0.019	30.5

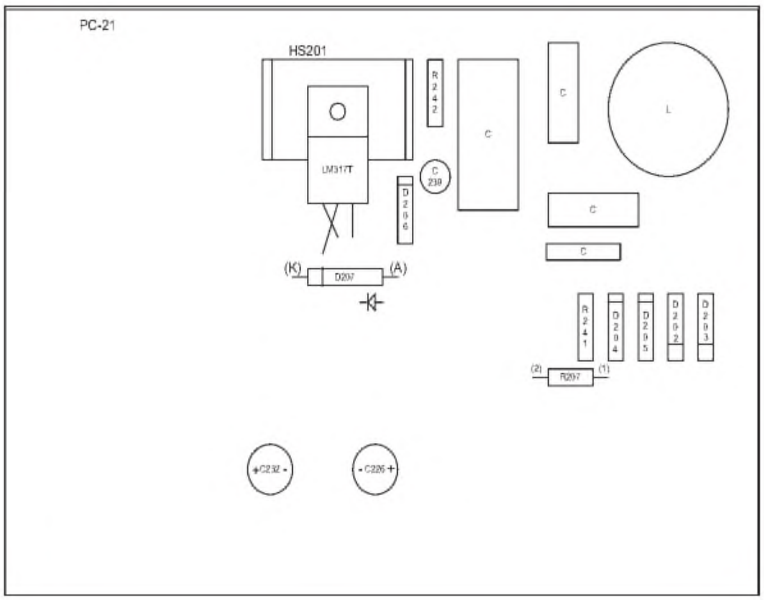


FIGURE 3: Principal parts placement on PC-21. A-2137-3

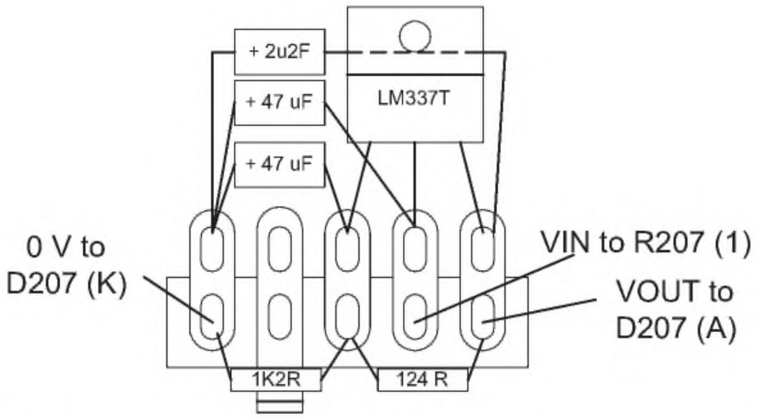


FIGURE 4: A guide for parts placement. A-2137-4

POWER-SUPPLY MODIFICATIONS

Safety first. DANGER! High voltage.

The area you will be working on is near the supply line power connections at either 120V or 240V AC. Do not work alone with high voltage present. When taking measurements always unplug, connect your test equipment, plug in, measure, and then unplug.

To proceed, unplug the unit and remove the top cover. Remove the two front-most back panel mounting screws and loosen the rear ones so you can fold back the rear panel. This way you can access the circuits, but the rear panel stays attached and the wires will not be strained and break.

If your tuner's power supply is working, connect your voltmeter to the positive supply, plug in, and turn on. Measure the supply voltage after about 20 minutes warm-up and record. The value should be about 13.2V to 13.6V. Unplug and repeat for the negative supply. Record the negative supply value.

Do not be tempted to raise the supply

TABLE 4
NEGATIVE SUPPLY DRIFT MEASUREMENTS

	ORIGINAL CIRCUIT VOLTS, DC	LM337 VOLTS, DC
Initial	NA	−13.2147
After 3 hours	NA	−13.2183
Total drift	NA	0.0036

TABLE 5
NEGATIVE SUPPLY, LINE REGULATION MEASUREMENTS

LINE VOLTAGE VOLTS, AC	ORIGINAL CIRCUIT VOLTS, DC	LM337 VOLTS, DC
105	−13.13	−13.2156
120	−13.33	−13.2174
130	−13.44	−13.2187
Regulation	0.31	0.0031

FM-5 FIX

While researching for this project, I read the initial FM-5 test reports from both *Popular Electronics* and *High Fidelity* magazines, reprinted by Dynaco. I noticed that the frequency-response graph showed a noticeable amplitude drop at the low end, −3dB at 30Hz or 40Hz relative to the 1kHz level. I decided to investigate.

The measurements I made confirmed the drop at 30Hz. Possible culprits are C219, C220, C229, C235, C225, and C231. I bypassed each of these with a capacitor ten times larger in value and measured. I found no measurable improvements.

Then I noticed C226 and C232. Computations show these to be the real source for the drop. I paralleled each with a 47µF 25V aluminum electrolytic capacitor and measured. The 30Hz drop was gone.

Figure 5 shows the simulation results of the amplifier circuit with different values for capacitor C226. The values used are 5µF (bottom left curve) and 47µF (top left curve). The PC-25 and PC-26 assemblies used in the later version FM-5 and the AF-6 have a different circuit and, as far as I can tell, do not suffer from this low-frequency drop.

Some will say that FM regulations limit the low end to 50Hz, but I suspect some stations are able to provide better low-frequency response than this.

To modify, unplug the tuner, locate C226 and C232. Connect the new 47µF in parallel observing proper polarity. Do not go above 47µF in value for the added capacitors. I do not know whether there is noise below 20Hz in this tuner, and I would not like to find out.

Figures 5 and *6* show simulated and measured low-frequency response, respectively. Distortion at 1V output is about 0.01% at 1kHz.

I reassembled the unit, and it works much better. Improvements are like onions, one layer at a time.

Alpha Transmission Lines

This author challenges some of the mystery and myth by using TSPs in transmission-line design. **By Rick Schultz**

Quarter-wave loudspeaker designs are mystical and mythical creations from the ancient past of loudspeaker design. They go back at least to Voigt's tubes (1934) and Olney's Acoustic Labyrinth (1936), about ten years after the invention of voice-coil drivers. Much later (1965), Bailey stuffed the entire pipe and called it a transmission-line.

Over the years, some people have built successful quarter-wave systems, but many more have failed. It is now possible to build a reliable α-transmission line derived from any driver's Thiele/Small parameters (TSPs). To get past the myths and mysticism, let's look outside the lines.

This design system takes some mystery out of transmission line (TL) construction. Later, I will cover how transmission line myth has hampered past designs. Even with this design system, the art of speaker building remains. You may decide to add or remove some stuffing as you listen to your α-TL. You may choose to build a longer or shorter α-TL for comparison.

TL DEFINED

A quarter-wave loudspeaker is a broad group of designs that rely on pipe resonance frequency to assist the driver's output. For example, when you blow across the top of an empty bottle, the bottle hums. If the bottle is partly filled, the hum sounds different.

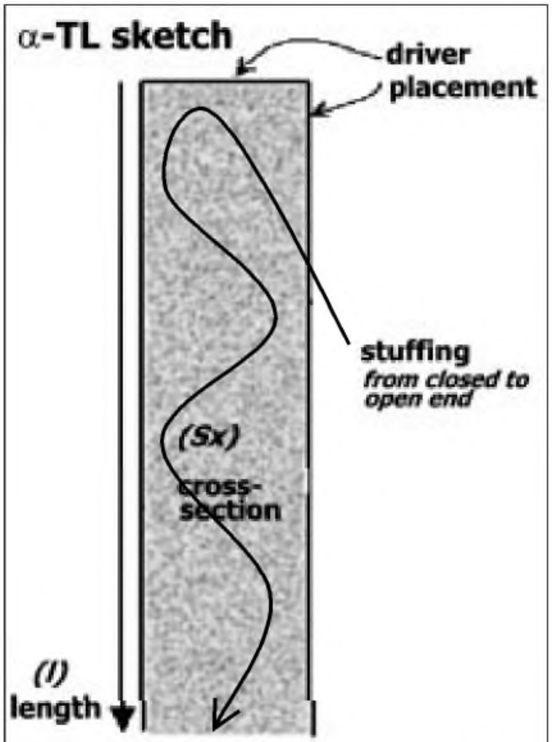


FIGURE 1: α-TL sketch.

The common element of quarter-wave design is a long narrow pipe, which is closed at one end and has an opening at the opposite end, just like the bottle. A driver is placed at or near the closed end. Most are stuffed with loose fibers.

Pipes and bottles with one end closed have unique resonant properties. An unstuffed pipe resonates at a frequency directly related to pipe length. A pipe of a certain length is often referred to by its quarter-wave frequency. For convenience, I label the pipe frequency fΘ. This resonance frequency is at the speed of sound divided by four times the length. I write this as $f\Theta = c/4l = 13560/(4 \times \text{inches})$.

Simply, if a closed pipe is 100″ long, its first resonance occurs at $13560/(4 \times 100) = 34$Hz. Interestingly, a 34Hz sound has the wavelength of 400″. The 100″ pipe is one-quarter of the 34Hz wavelength. Hence the name quarter-wave loudspeaker. (See the sidebar for more.)

TL DESIGN

This article focuses on transmission line design (*Fig. 1*). TLs are special quarter-wave systems. Specific TL properties are:

- Driver attached to the closed end of the pipe
- Opposite end of the pipe is completely open
- Internal stuffing fills the entire length
- Pipe is straight and not tapered
- Q-neutralized impedance curve
- Critically damped, transient perfect driver response

In an α-TL, the driver is critically damped every time. I have run many simulations, and driver rolloff is usually near 10dB per octave. Group delay is consistently quick and smooth. Traditional TLs perform very much like a $Q_B = 0.5$ acoustic suspension sealed box. Shorter α-TLs are also critically damped with great group delay, but offer additional low-frequency output. α-TLs are just as easy to build as a sealed box.

Here are the steps to design an α-TL for your driver.

1. Choose the pipe frequency fΘ (between f_S and two times f_S).
2. Decide whether you will use fiberglass stuffing, polyester, or AcoustaStuf.
3. Refer to *Fig. 2* or *Fig. 3* to read pipe length from the curve marked "length inches" line and left axis.

4. Refer to *Fig. 2* or *Fig. 3* to read stuffing density from the stuffing density line and right axis.
5. Refer to *Fig. 4* or *Fig. 5* to read the value of Z, at the same fΘ used in steps 3 and 4.
6. Calculate α equals $\sqrt{(1 + (f\Theta/f_S Q_{TS}))}$. Be careful to take the square root.
7. Cross-section area (Sx) equals $Z \times f_S \times f\Theta \times V_{AS}/(10{,}000 \times \alpha)$. V_{AS} must be in cubic inches.
8. Material needed equals length times cross-section times density.

Start by reviewing TSPs of the driver—any driver works. Important TSPs in this design system are resonance frequency (f_S), quality ratio (Q_{TS}), and equivalent volume (V_{AS}). One of my

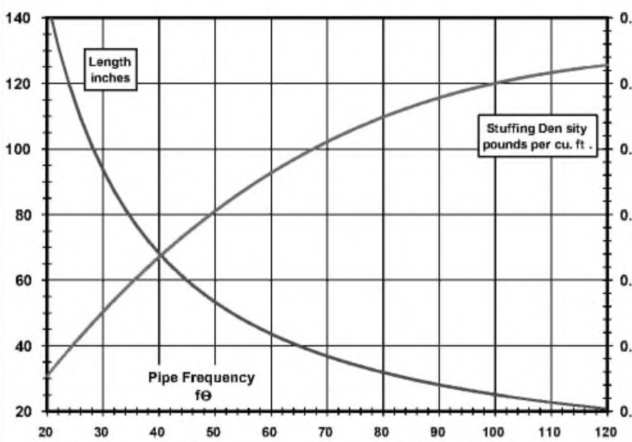


FIGURE 2: α-transmission line length and stuffing density—fiberglass. After you decide which pipe frequency fΘ you want to use for your project, use the length line to read the proper length. The length is in inches on the left axis. At the same fΘ, read the fiberglass stuffing density and the recommended stuffing amount on the right axis. Stuffing is given in pounds per cubic foot. To convert to metric units: inches divided by 39.35 equals meters. Pounds per cubic foot times 16 equals grams per liter. B-2215-2

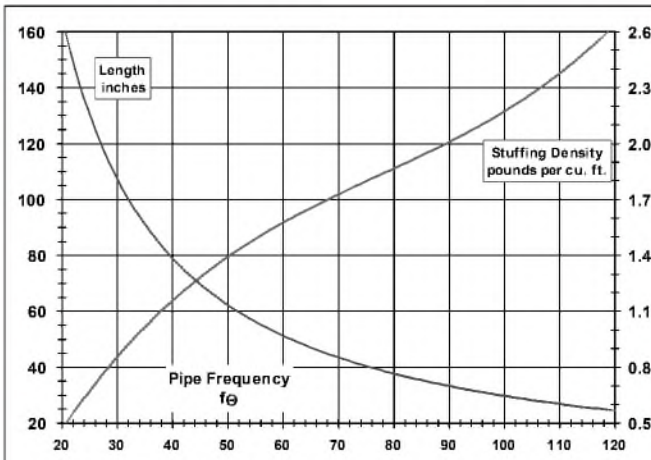


FIGURE 3: α-transmission line length and stuffing density—polyester, cotton, AcoustaStuf. After you decide which pipe frequency fΘ you want to use for your project, use the length line to read the proper length. The length is in inches on the left axis. At the same fΘ, read the fiberglass stuffing density and the recommended stuffing amount on the right axis. Stuffing is given in pounds per cubic foot. To convert to metric units: inches divided by 39.35 equals meters. Pounds per cubic foot times 16 equals grams per liter. B-2215-3

personal favorites is Jordan's JX92S (about $120). From Jordan's website at http://www.ejjordan.co.uk/92S.html, f_S is 45, Q_{TS} is 0.40, and V_{AS} is 15.28 ltr.

Notice whether V_{AS} is measured in liters (l), in cubic feet (ft^3), or in cubic inches (in^3). V_{AS} must be in in^3 to get proper results in my equation. If V_{AS} is in liters, multiply by 61.02 to convert to in^3. If V_{AS} is in cubic feet, multiply by 1728 to get in^3. For this driver, V_{AS} is 932 in^3.

Going through the steps:

1. I choose to set pipe fΘ = 80.
2. I will use fiberglass stuffing. I prefer the size and sound of fiberglass pipes, but polyester or AcoustaStuf is much easier to work with. I need to use *Figs. 2* and *4* for the next steps.
3. Referring to *Fig. 2*, my pipe must be 32″.
4. Refer to *Fig. 2*; the recommended stuffing density is at least 0.75 lb per cubic foot.
5. Refer to *Fig. 4*; the value of Z at 80 is 0.34.
6. α equals $\sqrt{(1 + (80/(45 \times 0.40))} = \sqrt{(5.44)} = 2.33$.
7. Cross-section area (Sx) = $0.34 \times 45 \times 80 \times 932/(10{,}000 \times 2.33) = 49\ in^2$.
8. Material = $32 \times 49/1728 \times 0.75 = 0.68$ lb or 11 oz. I need to divide by 1728 to convert in^3 to ft^3.

About 32″ is a very reasonable speaker height. If the fΘ you want to use requires a long pipe, there is more to follow about shorter TLs. The cross-section of this pipe is about 49 in^2. Cross-section shape of your TL is up to you. For the JX92S, cross-section could be a 7″ square, an 8″ diameter circle, or a 9 × 5½″ "golden ratio" rectangle.

Although you may consider this Jor-

QUARTER-WAVE LOUDSPEAKERS

Quarter-wave systems are a group of loudspeakers that rely upon the resonance of a pipe. The fundamental resonance of a quarter-wave pipe is based on the length of the pipe. I refer to this resonance frequency as fΘ.

The closed pipe frequency is expressed mathematically as fΘ = c/(4 × length), where c is the speed of sound. This is the same equation used to calculate one-quarter wavelength of a certain frequency. If length is in meters, speed of sound is 342m/s. If length is in inches, speed of sound is 13560 inches/sec. A pipe 42″ long resonates at 80Hz, the fundamental.

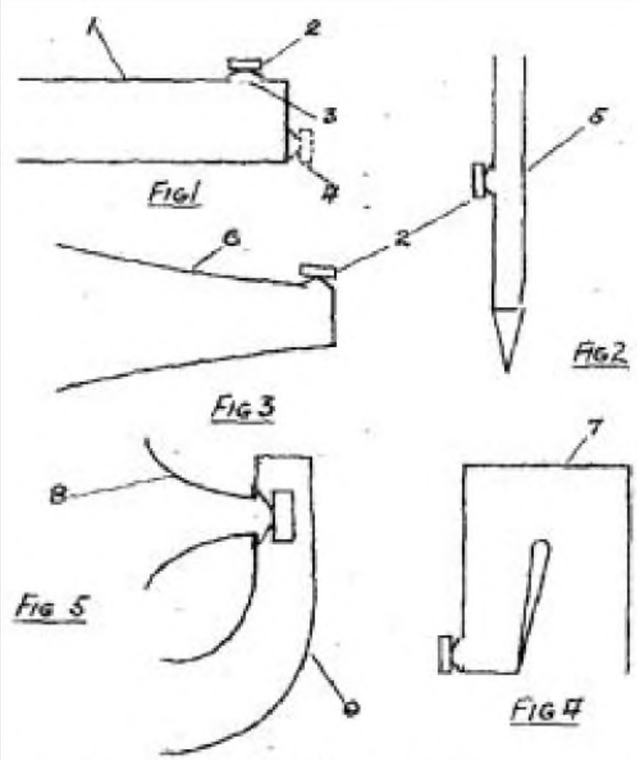
Unfortunately, bare pipes also resonate at odd multiples of fΘ (three times fΘ, five times fΘ, and so on, which I refer to as 3Θ, 5Θ, 7Θ, . . .) The 80Hz pipe will also resonate at 240Hz, 400Hz, 560Hz . . ., the harmonics. These extra harmonics are unpleasant and destroy the valuable sounds a loudspeaker is meant to deliver. Quarter-wave systems rely on stuffing to control unwanted resonances at 3Θ, 5Θ, 7Θ, and beyond.

Tapered Quarter-Wave Tube (TQWT)—The first quarter-wave system published was the Voigt (pronounced vote) Patent 447,749 submitted Oct. 17, 1934. In it he lays claim to a wide variety of quarter-wave schemes. However, nowhere does he mention fibrous stuffing. Without it, his pipes undoubtedly produce a series of unpleasant harmonics. His tapered design (*Fig. 3*) and folded tapered design (*Fig. 4*) have an ongoing cult-like following. Today, stuffing is always used in the tapered pipes.

Voigt describes his pipes as "closed at one end and excited by means of a loudspeaker diaphragm . . . [the] length just under one quarter of the lowest frequency at which efficient working is desired, the diameter being about ⅓ or ¼ the length, and the tube being closed at the one end." (Paul Gustav Adolphus Helmuth Voigt, Patent Specification 447,749, Application date: Oct 17, 1934.) His quarter-wave pipes place the driver on the closed end or on a side immediately adjacent to the closed end. He points out "in practice it is desirable to taper the bass chamber slightly . . ." TQWT is commonly referred to as the "Voigt pipe."

Acoustic Labyrinth—"It consists essentially of an absorbent walled conduit having one end coupled tightly to the back of the loudspeaker cone and the other end open. This conduit is in effect folded within the interior of the cabinet."[3] The photo accompanying Olney's article shows a pipe folded into three sections within a furniture-styled cabinet of that time. He places the driver at the closed end and the open end is near the floor.

Although no dimensions are given, the pipe appears to be several feet long and perhaps a half-foot square in cross-section. One side of the pipe is lined with thick felt, perhaps 2″ or 3″ thick.



From "Improvements in Means for Converting Electrical Energy into Sound"
Paul Voigt, U. S. Patent 447,749 (1934)
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"However, by lining the tube with material whose sound absorption rises suitably with frequency, we can suppress entirely these higher resonances . . ." Just two years after Voigt's pipes, Olney understood the importance of absorbent lining to suppress the 3Θ, 5Θ, and other harmonics.

Transmission Line—A long pipe with parallel sides, stuffed the entire length. Bailey writes more about the transmission line in 1972[4], "Radiation from the back of the driver cone flows down a pipe filled with a low-density sound-absorbing material. Fibrous absorbents such as loose wool, cotton wool, and kapok can be used." He also notes "[t]he effect of wool filling in the pipe is to slow down the wave relative to its velocity in free air." Stuffing not only attenuates the rear wave but also slows the speed of sound within the tangle.

dan JX92S example a large box, the amount of room it takes up on your floor is not that big, just 7 × 7″ plus the width of your wood. α-TLs have a very reasonable "footprint" compared to a corresponding sealed box. Going through the eight steps has given three critical bits of information: length, cross-section, and stuffing amount. Now you can build the α-TL. The sidebar describes construction. See *Fig. 6* for results of this 80Hz α-TL.

Since this is your first view of an α-TL, I need to point out some features in *Fig. 6*. Driver rolloff is near 9dB per octave. I have run many simulations, and the α-TL driver is critically damped every time. Though not on the graph,

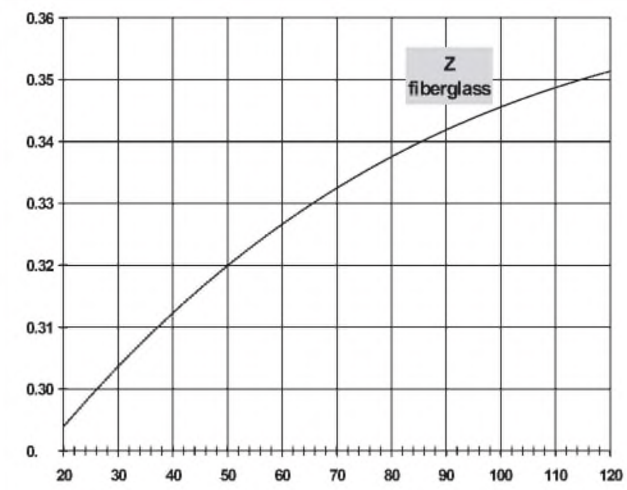

FIGURE 4: α-transmission line Z value—fiberglass. The Z value is needed to determine the cross-section area. Using the same pipe frequency, read the Z value from the curve. Use Z in this equation to set proper cross-section: Sx equals $Z \times f_S \times f\Theta \times V_{AS}/(10{,}000 \times \alpha)$, where α equals $\sqrt{(1 + (f\Theta/f_S \times Q_{TS}))}$. The shape of the cross-section is up to you.

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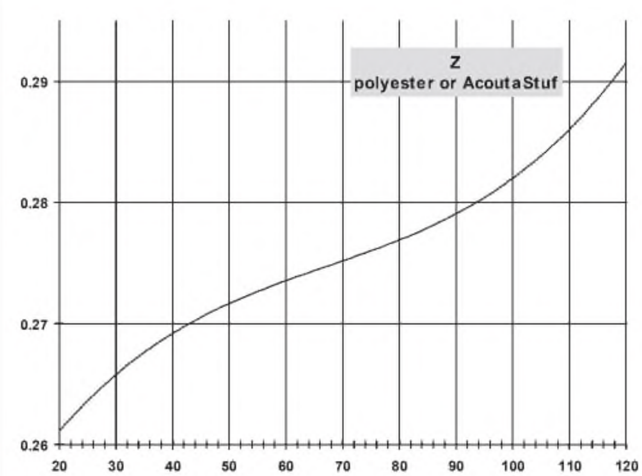

FIGURE 5: α-transmission line Z value—polyester, cotton, AcoustaStuf. The Z value is needed to determine the cross-section area. Using the same pipe frequency, read the Z value from the curve. Use Z in this equation to set proper cross-section: Sx equals $Z \times f_S \times f\Theta \times V_{AS}/(10{,}000 \times \alpha)$, where α equals $\sqrt{(1 + (f\Theta/f_S \times Q_{TS}))}$. The shape of the cross-section is up to you.

B-2215-5

group delay is consistently quick and smooth, normally between 4 and 6ms.

Notice how flat the impedance curve is in *Fig. 6.* α-TL impedance curves are noticeably flatter than the driver's free-air curve. Impedance curves of α-TLs are much flatter than the curve of the driver in a sealed box or bass reflex. If you like tube amps, this is an attractive property of α-TLs.

Lastly, output from the opening adds about 3dB to the low-frequency response of the driver. The low-frequency cutoff of the driver alone is at 93Hz. The opening output extends cutoff to 66Hz, one-half octave.

TRADITIONAL TL

Reviewing TL properties: they have straight sides; they are not tapered. The driver is always at the closed end of a TL. At the open end, the entire cross-section is open. Stuffing is uniformly packed along the entire length of the pipe, from closed end right to the open end. Traditional TLs have one additional, unique property:

- Pipe frequency fΘ is set near f_S

Let's examine a traditional TL, using the same JX92S for this example. The only prerequisite for making a traditional TL is setting fΘ equal to f_S. For the JX92S, fΘ = f_S = 45. Using fiberglass stuffing as before, *Fig. 2* shows the traditional JX92S TL is 60″ tall, while proper packing is just 0.57 #/ft^3. Doing the rest of the math, cross-section is 17 in^2, and 5.4 oz of fiberglass is needed to stuff the pipe.

Figure 7 shows the performance for this pipe. Notice the small output from the opening (dashed line). At 200Hz, opening sound is –21dB compared to the driver. That is 1/128 as quiet as the driver—very quiet. These traditional TLs attenuate the opening very effectively. They do not support and augment the driver. This confronts a TL myth. Lighter stuffing alone does little to improve bass response of the system.

As one other example of this, see *Fig. 8* of a traditional TL using the Peerless 2732, which is critically damped. At 200Hz, opening output is –26dB, or 1/400 driver output. Also notice how flat the impedance curve is for this traditional TL. Unfortunately, there is little low frequency support even though fΘ = f_S = 34.

STUFFING

Attenuation of the pipe depends on 1) stuffing density, 2) material, and 3) pipe length—not very profound! Tight packing means more fibers are encountered by the resonant sound waves. Longer pipes mean more fibers are encountered. Fibers of the material have their own unique diameter. Fiberglass is much finer than polyester or AcoustaStuf, has more fibers per unit volume, and is much more effective at suppressing pipe output.

In the first draft of this article, I presumed there was no standard density of stuffing. In his landmark article[1], George Augspurger provided recommended stuffing amounts appropriate for special TL geometries "but not for simple, straight pipes."

TABLE 1

Using the Jordan JX92S, pipe resonance may be set at any frequency. This table shows cabinet size and stuffing for three resonances and two materials. Results for these values are compared in *Figs. 9* and *11, Figs. 10* and *12.*

JORDAN JX92S FΘ VARIATIONS	LENGTH	DENSITY #/FT^3	CROSS-SECTION	OUNCES OF MATERIAL
FIBERGLASS STUFFING				
45Hz (f_S)	60 in	0.57	32 in^2	10.2 oz
63Hz (1.4f_S)	41 in	0.68	41 in^2	10.7 oz
90Hz (2f_S)	28 in	0.78	53 in^2	10.8 oz
POLYESTER STUFFING				
45Hz (f_S)	70 in	1.28	27 in^2	23 oz
63Hz (1.4f_S)	49 in	1.63	35 in^2	26 oz
90Hz (2f_S)	34 in	2.01	45 in^2	28 oz

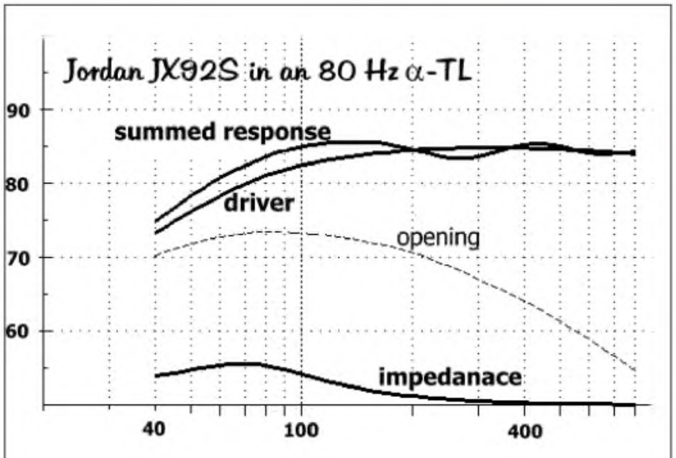

FIGURE 6: Response and impedance curves of the Jordan JX92S in an α-TL. This pipe is designed at 80Hz. Notice the broad flat impedance curve centered at about 70Hz. The sound produced by the driver is the bold line. As you follow it from midrange down to low frequencies, the slope is –6dB per octave. This indicates critical damping. Critically damped systems have excellent transient response. This is described as "clean" or "tight" sound. See the text for a brief discussion of the "summed response" curve. B-2215-6

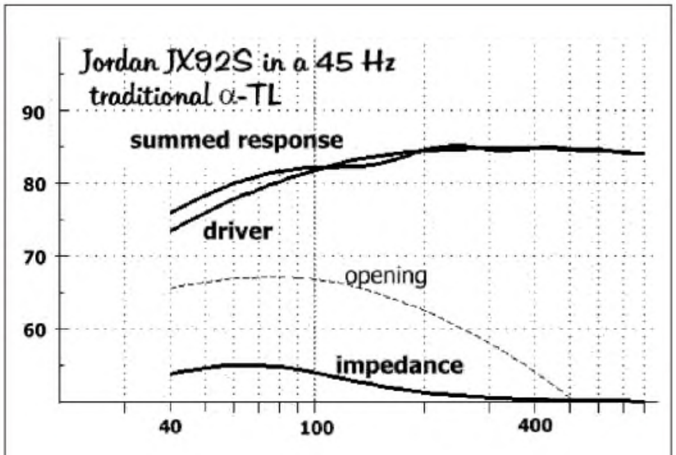

FIGURE 7: Traditional α-TL response. The pipe is constructed so pipe resonance is the same as the JX92S resonance 45Hz and stuffed with fiberglass. Compared to *Fig. 6*, the impedance curve has a similar shape but is centered at 64Hz. Driver response is identical. However, the longer pipe with lighter stuffing has less opening output than the 80Hz pipe in *Fig. 6*. In the past, it was assumed a longer pipe produces more bass. In fact, longer pipes are quieter at low frequency than short pipes. B-2215-7

As I reworked this piece, I looked at the right amount of stuffing for a given straight pipe frequency. It does not matter what driver f_S, Q_{TS}, or V_{AS} is. It does not matter how wide or slim the pipe is. The density of stuffing depends only upon fΘ. When determining the right density, I looked for about a 1.0dB dip at the 3Θ resonance and less than 0.5 lift beyond 3Θ. As I re-read his article, I realized Augspurger used this same standard.

Voice-coil drivers show more variation than this. When cabinet output meets this standard, the α-TL cabinet offers less coloration than the driver itself. That is quite a bold statement. My point is the driver is rarely uniformly flat through midrange. α-TL cabinets show less variation than the driver itself, but this is not the first time this claim has been made.

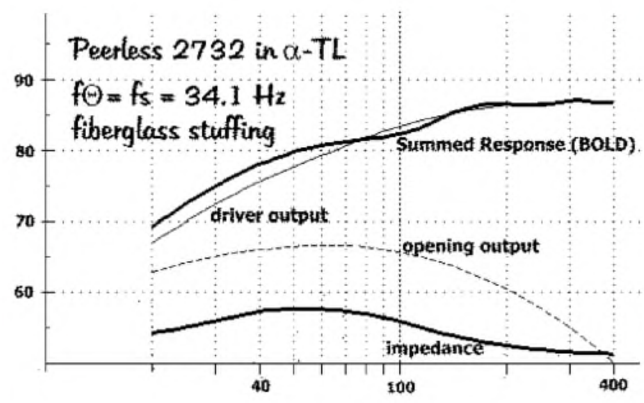

FIGURE 8: Traditional α-TL using the Peerless 2732. This graph is another example of how quiet long pipes are. The pipe frequency is the same as driver resonance, which is 34Hz. Summed response quiets noticeably below 125Hz. The shelving of response between 50 and 110 is largely due to the 3Θ resonance. When 3Θ resonance is better controlled, the shelf would be smoothed out. (Thanks to Peerless, who provided these drivers for an earlier, failed project.)

B-2215-8

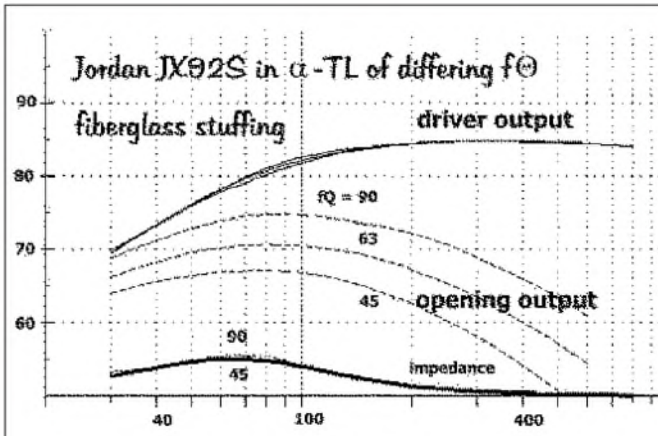

FIGURE 9: α-TL summed pipe responses—fiberglass. The fine lines on this graph are the combined output of the driver and opening seen in *Fig. 11*. This sums the amplitude and phase of the sounds and assumes the driver and opening are the same distance from the ear. The flattest response comes from the 63Hz pipe. High and low spots in the summed curves are due to phase differences between the driver and opening.

B-2215-9

"Listening tests proved that the [TL] cabinet had a 'cleaner' sound than the bass reflex type, the effect of the line being very noticeable in its lack of coloration on speech. Transient response was definitely better on the line speaker, the sound being more 'tight' and natural.

"The final subjective tests were very good. The sound quality is effortless

CONSTRUCTION

The first part of this Jordan JX92S project involves cutting the sides. To keep this project simple, I built a square cross-section. Start with four boards 31⅞″ long (*Photo 1*). MDF is the best to work with. I used ¾″ material for most of my projects because it doesn't split as easily and provides enough surface for adhesive. The inner dimension of the Jordan JX92S project is 7″ square. I cut the boards a little wider than 7¾″. This photo shows the driver opening layout.

PHOTO 1: All sides of this project are the same. Since the material is ¾″ thick, the side boards are just over 7¾″. The driver needs an opening of radius 2¼″.

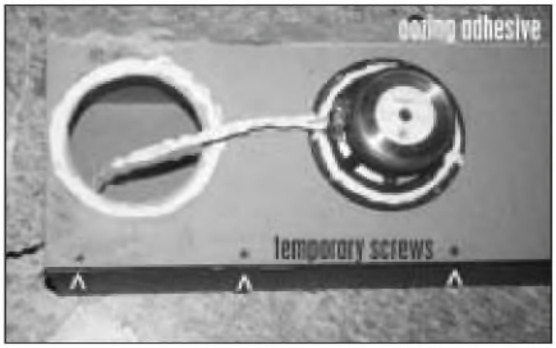

PHOTO 4: Assembled sides are held together with construction adhesive and temporary screws. The adhesive fills in gaps. The screws are removed the next day.

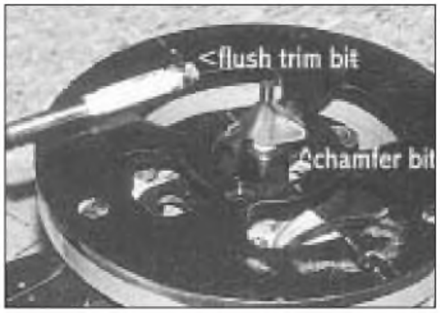

PHOTO 2: The chamfering bit used to widen the inside of the opening. Resting on the router is the flush trim bit used after assembly.

PHOTO 5: Bottom of the assembled cabinet. Extra blocks are added at the corners to receive the spikes. The round black object is the speaker wire d-cup.

PHOTO 3: The face board with opening cut out.

PHOTO 6: The finished α-TL featuring Jordan's JX92S. Since I had wallpaper available, which matches the living room walls, I tried a different "finish."

Before assembling the sides, I cut the driver opening. It is 4½″ diameter. I don't have a nice router jig, so I cut the opening free-hand. I used a plastic lid from a coffee can as my compass, which was very stable and made a perfect circle. I cut the hole using my saber saw, but I didn't get a perfect circle (*Photo 2*). If the driver doesn't fit, you can cut a little more from the hole. Often, I simply tilt the jig saw a little and just knock off the rim until it fits.

To me, the most important cabinet detail for smaller drivers is chamfering the rear of the hole. Chamfer is a 45° bevel cut. This is critical if you use thick material, which can obstruct airflow between the hole and basket, thus altering all the TSPs. Also, sound ricochets off a square-cut hole wall producing awful reflections back onto the cone.

The chamfering bit (*Photo 3*) allows free airflow from the rear of the driver and fewer reflection problems. Resting on the router is an edge trim bit (about $18). It is used later to square off and straighten the cabinet edges. A carbide edge trim bit will last for a few projects.

The sides are butt-joined and held temporarily with drywall screws. I pre-drilled the face but not the side and applied a continuous bead of Liquid Nails construction adhesive to one board. When I drove the screw straight into the material, I noticed some splitting of the side. I needed to reset some screws so I drove them on an angle without a pilot hole. This produced much less splitting, so I plan to use this angling technique in the future.

I used the Liquid Nails to glue four ¾″ × ¾″ × 2½″ blocks at the bottom inside corners. I ran some Latex caulk around the driver and the opening (*Photo 4*) and pressed the driver firmly into the caulk so it oozed out the front and rear. I installed four drywall screws along each edge. I was careful so I didn't over tighten them and strip the MDF, and let the assembly cure overnight.

The next day I removed the screws. With a utility knife I trimmed away the cured adhesive and caulk (*Photo 5*). Because I cut the MDF a little wider than needed, I trimmed the long MDF edges with the bit shown in *Photo 2*. The edges were straight and sharp.

At the end near the driver, I lined the inside of the cabinet with fiberglass ceiling tile, which was very effective at reducing midrange and treble reflections. I installed 10.6 oz of fiberglass throughout the cabinet. I cut the top MDF a little big, not quite 8 in^2, attaching it with Liquid Nails and pressing it into place. I didn't bother with screws. Later, I trimmed this with the router just as I did on the long edges. I installed four spikes at the bottom corners to keep the cabinet off the ground. These spikes were essential to detailed imaging.

I used a different cabinet covering on the project (*Photo 6*), having some wallpaper left over from our living room. My first attempt with this pre-pasted paper did not work. To wallpaper an MDF cabinet, first seal the wood with wallpaper primer (also called sizing). Apply wallpaper adhesive to the paper and the cabinet. Clay adhesive is the best. Place the paper centered on the front and wrap around both sides.

I pre-cut the driver hole in the wallpaper and I think that was a good idea. Wipe out air bubbles with a soft rag or brush. Apply pieces to the sides and wrap to the back. Trim the rear, top, and bottom with a razor-sharp tool. Lastly, apply the top piece and trim. Wallpaper is very forgiving and covers minor imperfections such as unfilled screw holes.

and natural. At first hearing the bass sounds deficient but extended tests show that this is not so, it is merely that one has been conditioned to hear resonant bass."[2]

I must caution that every driver has its own properties. Some are better than others in an α-TL. If you notice some irregularity in the midrange, simply install a little more stuffing. You can remove a bit of stuffing to see whether you can get a little more low end. α-TL construction is still an art. This α-TL design system gives you a very good head start on your project.

You are encouraged to vary the amount of stuffing until you attain the sound quality you prefer. Heavier stuffing brings the system closer to the Q_B = 0.5 acoustic suspension. Packing lighter than my recommended density allows more reinforcement and cancellation, producing uneven response.

The curves in *Figs. 9* and *10* are the sum of phase and amplitude of the driver and opening. The sum ignores floor lift, the 2–3dB additional amplitude, because the opening is at the floor. The model assumes driver and opening are equidistant from the ear. However, the driver is often placed near ear level while the opening is farther away at floor level. Room reflections and ceiling height are ignored in the graphs.

For these reasons, I do not fully accept summed response above 120Hz.

The only way to tell whether your combination of stuffing material, density, length, and driver pleases you is to build an α-TL and listen to one in your room. Give the system some time and add stuffing to suit your taste.

FΘ—PIPE FREQUENCY

TL myth assumes you can lower system response by building a longer pipe; pushing fΘ lower must extend bass performance. This myth started 65 years ago when Voigt set "length just under one quarter of the lowest frequency at which efficient working is desired." He does not relate quarter-wave pipe frequency to driver resonance. Instead, he related it to whatever frequency "is desired."

Speaker builders have assumed just building a longer pipe will extend response. In fact, building a lower fΘ pipe just quiets it. Longer TLs perform much worse than shorter TLs. The lower limit for pipe frequency should be fΘ equals f_S. I recommend going no lower, any longer.

Shorter pipes produce +2 to +4dB additional bass through low frequency rolloff. This lift continues to near two times fΘ. Shorter α-TL achieves results similar to Augspurger's special geometries. As he comments on his geometries, he observes, "The efficiency matches that of an equivalent closed-box system; however, pipe output contributes 2–3dB in the low frequency range. . . . the net result is a corresponding increase in maximum output."

Augspurger also says, ". . . In contrast to a basic cylindrical pipe, at least four alternative geometries allow lighter damping, which results in higher efficiency." In fact, straight pipes can perform nearly as well as his geometries. The secret is higher fΘ and tighter stuffing.

Let's consider different pipe frequencies and their performance. I will use frequencies fΘ = f_S, $1.4f_S$, and $2f_S$ for contrast. Using the Jordan JX92S, the three pipe frequencies are 45, 63, and 90Hz. *Table 1* contains the cabinet dimensions using fiberglass and polyester. Fiberglass results are in *Figs. 11* and *9*; polyester results are in *Figs. 12* and *10*.

In *Figs. 11* and *12*, driver response is

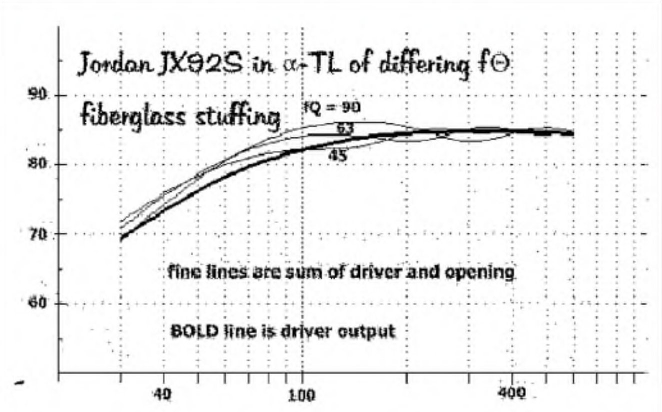

FIGURE 10: α-TL summed pipe responses—polyester. The fine lines on this graph are the combined output of the driver and opening seen in Fig. 12. This sums the amplitude and phase of the sounds and assumes the driver and opening are the same distance from the ear. These summed responses are very similar to the fiberglass results in *Fig. 9*. B-2215-10

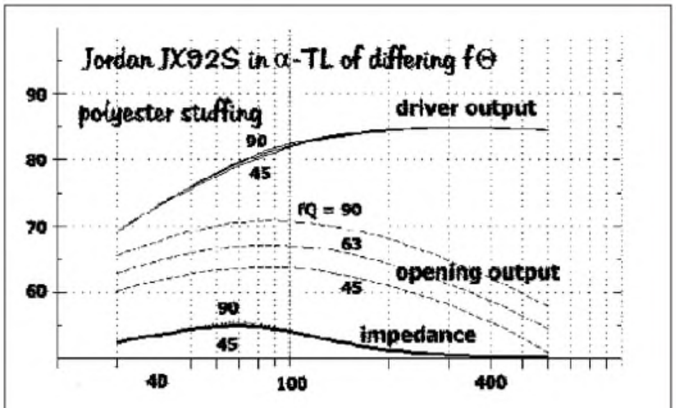

FIGURE 11: Comparison of three fiberglass stuffed α-TLs at different pipe resonances. Impedance is the same for all three pipes, driver output is virtually identical. The only difference is opening output. The longer lower pipes are clearly quieter. I recommend building no lower, any longer. B-2215-11

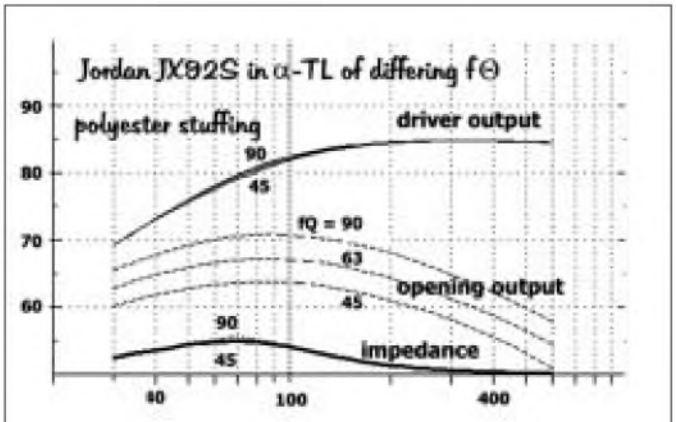

FIGURE 12: Comparison of three polyester stuffed α-TLs designed at different pipe resonances. Impedance is nearly the same for all three pipes; driver output is virtually identical. The significant difference is opening output, which needs to be quieter with polyester, cotton, and AcoustaStuf, so the summed response is appropriate (*Fig. 10*). B-2215-12

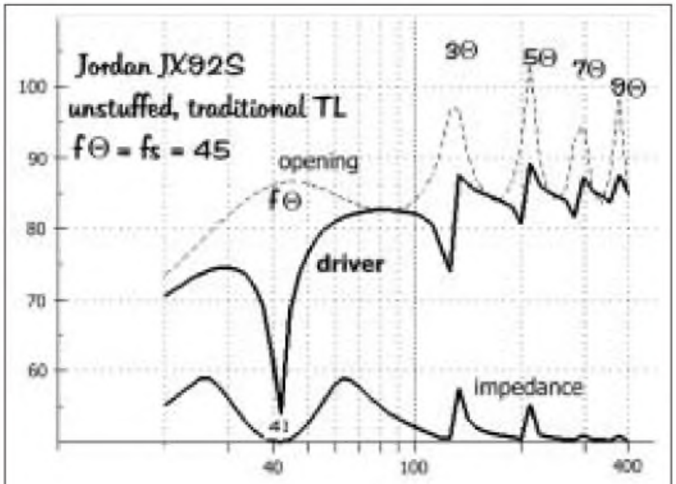

FIGURE 13: Unstuffed traditional α-TL response. The Jordan JX92S in an empty 45Hz pipe shows typical results. Augspurger's program makes numerous calculations and connects the points on the curves. The result is some clipping of the 3Θ and 7Θ harmonics. If this pipe was actually tested, the 3Θ and 7Θ peaks would be closer to the location of the labels. Notice the impedance anomalies at each harmonic. Also, opening resonance is at 45Hz but the driver is completely quiet, dead at 41Hz. B-2215-13

virtually the same—critically damped. Impedance curves are quite flat and practically the same. The only noticeable difference is opening output. The shortest pipe gets closest to driver output. However, there can be a lot of lift into midrange. I recommend choosing fΘ to be 1.2 to 1.6 times driver f_S.

Since length, density, and stuffing vary, cutoff is difficult to predict accurately. Cutoff is at least as good as the Q_B = 0.5 AS and may be as much as one-half octave lower, especially if fΘ is 1.2 to 1.6 times f_S. These pipes also avoid excessive midrange lift. Not bad considering the cabinet is smaller than a traditional TL.

IMPEDANCE

Pipe frequency is the primary reason the impedance curve flattens. There is a down side. Unstuffed TLs do not resonate at only fΘ. They have acoustic and electric harmonics at odd multiples of fΘ at 3Θ, 5Θ, 7Θ, and so on.

Figure 13 shows these ugly peaks for an un-stuffed traditional 45Hz pipe using the JX92S. These are unbearable and must be suppressed. The only solution is stuffing, which is vital to deaden acoustic and electric irregularities produced by pipe harmonics.

The basic impedance curve for the JX92S on a large flat board is curve A in *Fig. 14*. This curve is centered at 45Hz, so we say f_S = 45. The broadness of the curve determines Q_{TS}, which is 0.40 for this driver.

Next, look at curve C, which is very irregular. It is the curve of the JX92S in an unstuffed traditional TL of fΘ = f_S = 45. I set the cross-section based on my α-TL model. But this curve is more regular than it seems.

One important detail to notice is the first two peaks. If you consider the first two peaks alone, the curve looks like an exquisitely tuned bass reflex design. Amplitude is the same for these two peaks. An improperly tuned pipe exhibits unbalanced peaks.

The next important observation is the location of the peaks at 27 and 64Hz. These are important because f_S = 45 is very near the (geometric) center frequency of these two points; that is, $\sqrt{(27 \times 64)}$ = 41. Unstuffed α-TLs always display this centering. The center frequency is normally about 10% lower than f_S.

The third peak is near 135, the next near 225, another near 315, and again near 405. These impedance peaks represent the 3Θ, 5Θ, 7Θ, and 9Θ pipe harmonics. As these resonances quiet at higher frequencies, the impedance peaks flatten. Actual harmonics are normally just below the predicted harmonics.

Finally, curve B is for the traditional stuffed α-TL. Notice the broad smooth curve. Although the α-TL is tuned to 45Hz, the impedance peak is not at fΘ = f_S = 45. Stuffing deadens the first peak. Stuffing also deadens the pipe harmonics. The only remaining peak is at 64Hz, the same as the second unstuffed peak. Because the graph is on a logarithmic scale, this peak is only about 40% of the driver on a board.

So, it is pipe frequency that damps driver resonance; then stuffing damps pipe resonances. Proper pipe length and stuffing produce a flat impedance curve, phase unity, excellent transient response, and quick, uniform group delay. As fΘ moves away from f_S, cross-section adjusts so Q_{TS} is nearly neutralized in all α-TLs.

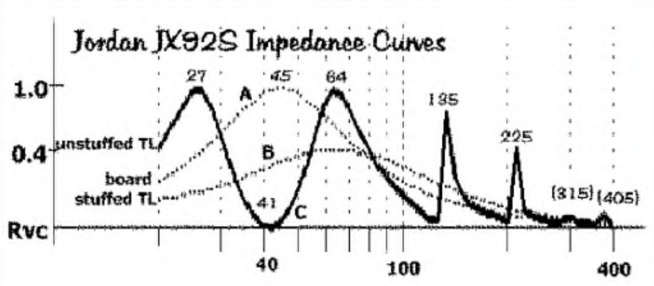

FIGURE 14: Comparison of unstuffed and stuffed traditional α-TLs. Line A is the impedance of the JX92S on a flat board. Line B shows the driver in a stuffed traditional α-TL. Line C is the wild impedance of an unstuffed pipe. Stuffing deadens all but one of the unstuffed peaks. The only remaining peak is located at the second unstuffed peak at 64Hz. Because the graph is on a logarithmic scale, this peak is only about 40% of the driver on a board. *Figure 14* holds the key to improving the α-TL. Low frequency response can be extended when the first and second peaks are developed and the third peak is subdued. B-2215-14

αLPHA . . .

Those of you interested in engineering loudspeakers may have noticed a loudspeaker mystery unveiled here. Speaker builders are accustomed to α associated to acoustic suspension cabinet volume. In all quarter-wave systems, α is length dependent. More accurately, it is dependent upon the $f\Theta$:$f_S Q_{TS}$ ratio. The α-ratio was the ultimate mystery in TSP-based TL design.

Properly designed quarter-wave systems neutralize the driver resonant peak defined by f_S and Q_{TS}. α-TLs compensate or flatten the driver's electrical resonance peak. When you look at the α formula, note what happens when $f\Theta$ is set equal to f_S. Then $\alpha = \sqrt{(1 + (f_S/(f_S Q_{TS})))}$ or $\sqrt{(1 + (1/Q_{TS}))}$. α is related to the inverse of Q_{TS}. This explains the flattened impedance curve at driver resonance.

To get the flattest possible impedance curve, simply set $f\Theta = f_S/Q_{TS}$. This does not ensure extended low frequency; it only produces the flattest impedance. To get both the best response and impedance, select a driver whose Q_{TS} is between 0.5 and 1.0. Still, many people prefer the sound of low Q_{TS} drivers (below 0.4) in their TLs. This design works for all speakers no matter what Q_{TS} the driver has.

This article and design is subject to full copyright and patent protection. There is no restriction, of course, on private individuals making an α-TL for their own pleasure. These are the (proprietary) mathematical equations for the curves on *Figs. 2* and *4*, *Figs. 3* and *5*. These equations are reliable for pipe $f\Theta$ up to 120Hz. Beyond that, other factors influence results.

$L_f = 3816(f\Theta\text{^}-1.09)$

$D_f = 0.0000002\,f\Theta^3 - 0.00009f\Theta^2 + 0.014f\Theta + 0.11$

$Z_f = 0.00000008f\Theta^3 - 0.00002f\Theta^2 + 0.0023f\Theta + 0.25$

$L_p = 4014(f\Theta\text{^}-1.06)$

$D_p = 0.000003f\Theta^3 - 0.0007\,f\Theta^2 + 0.068f\Theta - 0.64$

$Z_p = 0.00000005f\Theta^3 - 0.000009f\Theta^2 + 0.0008f\Theta + 0.25$

. . . AND ΩMEGA

Mr. Augspurger provided the technical review of this article. I deeply appreciate his interest and support. He comments, "Four things I learned from reading the revised manuscript: (a) A simple quarter-wave damped pipe is as easy to build as a closed box, but can provide certain performance advantages. (b) With straight pipe geometry, enclosure volume may not be all that important . . . it doesn't make much difference whether the pipe is 8 in^2 or 10 in^2. (c) The quarter-wave pipe frequency does not have to match the speaker's free-air resonance. For best performance, it should not. (d) To get more bass, make the pipe fatter, not longer. I agree with all four assertions." He also agrees with my assertion that there is a specific stuffing density dependent upon pipe $f\Theta$ and material only.

Impedance curves can be manipulated by cabinet modifications. The second peak is moved near f_S and the third peak is deadened by the cabinet and not the stuffing. Simple modifications to α-TL cabinet design and stuffing lead to a different TSP-based design—the Ω-quarter wave reflex. ❖

ACKNOWLEDGMENTS

I am deeply indebted to G. L. Augspurger, who generously provided his software. Using it has allowed great progress in pursuing my hobby. The graphs in this article are produced by his program.

For more on quarter-wave design, see the best QW site on the planet at http://www.t-linespeakers.org/index.html. Dave Dlugos runs the site and has been very supportive of my design ideas. This article will be posted there after publication.

Martin J. King has been very encouraging as we corresponded about stuffing, TLs, TQWTs, and exolinear design. Visit his informative site at http://www.quarter-wave.com

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System Power Control

Try this circuit to control the power to your audio system.

By Frank S. Thomas III

After installing a new preamp in my system, I experienced audible, but not dangerous, thumps upon turn-on and turn-off. The preamp incorporates tightly regulated power supplies, while the power amps incorporate the usual capacitor input filter power supply. Neither of the power amps—a Hafler DH200, highly modified, nor the new Hsu subwoofer amp—had ever caused any turn-on or turn-off artifacts. The problem was in the preamp power supply.

I decided to solve the problem with a power control unit, even though another solution is to simply manually switch the power amp(s) on after switching on the preamp; and switching the power amp(s) off before the preamp. Most of the time I remember this unusual sequence, but other members of my family do not. I determined that a time delay of five seconds was enough to allow the preamp power supply to stabilize before switching the power amps on or off.

However, this imposed an unusual requirement. I know of no reasonably priced time-delay devices that turn on in one sequence and turn off in the opposite sequence. You could make a logic system work, but I am not very skilled in designing such circuits.

Furthermore, the power levels would require both voltage and current translation from the logic circuit levels. So I decided to combine a simple time-delay device with appropriate switching to control the three AC outlets. After several attempts, I developed the circuit shown in *Fig. 1*.

CIRCUIT OPERATION

The time-delay device is an ancient Amperite thermal relay (TD) with a nominal five-second delay at turn-on, a slightly longer delay at turn-off, and a 115V AC heater. However, I didn't want to switch the AC to the power amps through the Amperite relay contacts. In this design, the TD powers a transformer/rectifier circuit to energize a power relay, which you can select to match the current requirements of the main power amps. You may use another type of TD, provided it has sufficient delay both at turn-on and turn-off.

There are two additional requirements for this circuit to work properly. The first is that the main switch must be a DPDT type. One pole supplies AC to the TD relay heater, while the second supplies AC to the relay contact/transformer/rectifier/power relay. They can't be combined into one switch contact or the circuit will "lock up" and won't turn off, due to the hold-in action of the preamp power circuit from the power relay.

The second requirement is that the preamp power must be maintained until the TD relay drops out. Remember, the preamp turns on first and turns off last. This is accomplished by providing AC from the main switch for turn-on and from one of the power relay contacts to maintain AC power until the TD times out.

I designed this circuit to power my system. The preamp, which appeared in *Stereophile* magazine, November 1991, as "Aunt Corey's Homemade Buffered Passive Preamp," is a homemade almost passive unit made popular by Corey Greenberg. It is powered by a separate, very robust, power supply described by Gary Galo in *The Audio Amateur* 4/90. It is housed in a surplus PAT-5 chassis, with a custom smoked glass front panel, and gold knobs from an earlier Dynakit. It incorporates a separate headphone amplifier from Headroom.

Two main amplifiers are used: a Hafler DH-200, much modified, driving biwired Vandersteen 2Ci main speakers, and a stock Hsu Research driving a single Hsu TN1220H0 subwoofer. Source components are a Marantz CD-

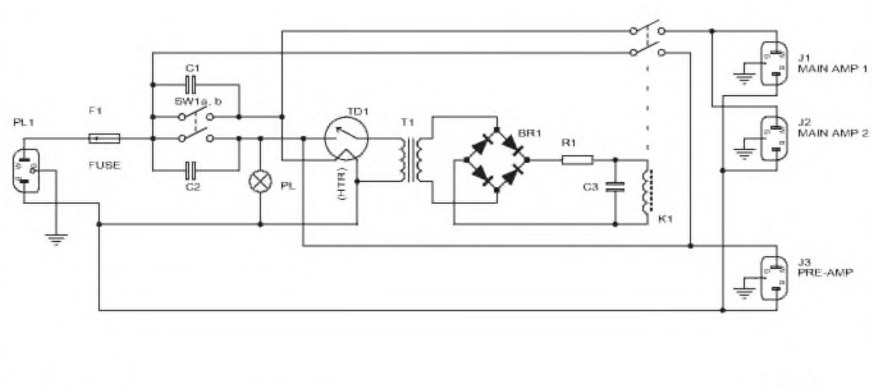

FIGURE 1: Power control circuit.

63 CD player, with digital output to a separate Assemblage DAC-2 digital processor. The FM tuner is a Dyna FM-5 rebuilt into a Super Tuner Two by Frank Van Alstine. Interconnects are homemade using Mogami cable. Future plans include a new equipment cabinet and new speaker cables.

ABOUT THE COMPONENTS

The main fuse is a compromise between adequate capacity to carry the main amp currents and protection of the start-up system elements. I used a 20A fuse, which provides little protection in the event of a failure of T1, BR1, or the K1 relay coil. In over 50 years I have never had a failure of this type of component. Nevertheless, you can add a second fuse to protect these components just before the TD1 switch contact, but after the junction point which feeds the preamp receptacle. This fuse can be in the range of 1–2A.

The DPST switch is a 20A toggle style. This may be difficult to find, so other styles or a common two-pole household wall switch rated for 20A will work just as well. C1 and C2 are 0.01µF, 1000V disc capacitors, which are connected to the line and should be so rated for this application. PL1 is a neon pilot light rated for 115V, but an LED with dropping resistor would work just as well. TD1 is an Amperite octal base time-delay relay with a 115V heater and a five-second time delay. The contact rating is 3A.

T1 is a small, 12.6V secondary transformer. A current rating of 300mA is sufficient for the power relay I used, so a Radio Shack 273-1385 or 273-1365 will be more than adequate. However, if you decide to use one or more heavy-duty relays, you may need to substitute a Radio Shack 273-1352. BR1 is a 100V, 1A bridge rectifier. A Radio Shack 276-1161 is suitable.

To provide current limiting to the power relay, R1 is a 150Ω, 2W resistor. You will need to adjust the resistor value to suit both the transformer output voltage and the power relay coil current. C3 is a 100µF, 50V electrolytic. K1 is a Radio Shack 275-218, DPDT with 10A contact rating.

The coil current in the current catalog is 130mA at 12.6V; however, the one I used was listed at 75mA, which sized the power rating for R1. It is a plug-in style, but I mounted it upside down and soldered wires directly to the lugs. As I discussed earlier, if your main amplifiers require more current capability, a 20A DPDT relay with a 12.6V coil is available from Radio Shack Unlimited, as part number RSU 12131496.

I did not include RFI/EMI filtering since each of my components includes it, and the power control is fed from a filtered AC supply. The two major options are the TD and the power relay. You can use a different TD, provided it has sufficient turn-on and turn-off time delays. Also, you can eliminate the transformer/rectifier-power relay if the TD relay has sufficient current capability, and has separate switch contacts to supply the main amp power separately from the preamp power, just as the power relay does in this design.

The power relay I used has a 12V DC coil, because I didn't want any AC buzz from a 115V AC coil, but you could experiment with it and eliminate the transformer/rectifier. Ideally, the power relay could be a 3PDT so each power amp could be fed from a separate contact. The DPDT I used has 10A contacts and could be considered marginal. Other relays are available from Radio Shack with 3PDT, 10A contacts and DPDT, 20A contacts by special order. If overkill is to your liking, RS also has 30, 40, and 70A contact relays, but they are SPST.

The resistor is used to control the power relay coil current, and its value will depend on the selected relay and the transformer/rectifier output. The capacitor simply provides some ripple smoothing. Obviously, the main fuse, power switch, and pilot light are selected to provide sufficient capacity. The capacitors across the power switch may not be necessary, but I like to use them. I built the unit in a 4 × 5 × 6″ mini-box, but if some of the options are chosen, a smaller box could be adequate.

The unit has been in my system now for about two years and is working just fine. Readers with any feedback or alternative bright ideas can reach me at franktren@sbcglobal.net. ❖

Build a Reactive Load

This author builds a reactive load to test the behavior of his amp designs. **By Dick Crawford**

PHOTO 1: The finished box.

On page 66 of the May 2002 issue of *audioXpress*, Dennis Colin writes:

"We all know how reactive most speakers are, but consider how profoundly this exacerbates amp(lifier) distortion: you have a wildly-gyrating elliptical load-line plunging the output circuitry into all kinds of normally uncharted regions (sounds figurative but is literal here). The 'wild gyrations' are from the vast multitude of simultaneous musical tones exciting the usual speaker impedance variations (magnitude and phase) across the audio band."

This struck a chord, as I've sometimes wondered how my amplifier designs would behave using a reactive load. I decided to design and build a reactive load . . . but what kind? Mr. Colin advises both inductive and capacitive loads at 4 and 8Ω, and at 30 and 60° phase angle, for several different frequencies. This would represent a closetfull of reactive loads, and my closets are already full of audio junk.

So I decided to build the worst-case reactive load: inductive and capacitive at 60° over a range of audio frequencies. I chose 4Ω for this design. I found that I could combine a resistive load, an inductive load, and a capacitive load all in one unit (*Fig. 1*).

COMPONENTS

The resistive load is the heart of this design, with additional circuits representing the inductive and capacitive segments. Since the 4Ω resistive load is in series with the other loads, the load on the amplifier is never less than 4Ω. This is only fair, as some amplifiers (even good ones) do not like loads of much less than 4Ω.

The inductive load is about 3mH, which could be a big subwoofer. The capacitive load is about 1μF, which could be an electrostatic driver. The finished box (*Photo 1*) has binding post (banana plug) connectors, with the usual ¾″ spacing, with one common binding post (black) and three binding posts (red) being the inductive, resistive, and capacitive connections.

Because I did not want the components to add measurable distortion, I used air core inductors, metallized polypropylene capacitors, and wirewound resistors for this design. If you have some other preferred components, please use them. Just remember Crawford's edict, which says that all parts have some distortion.

PERFORMANCE

My goal was reactive loads of 60° (±10°). *Figure 2* shows the measured reactive loads. The inductive load achieves specifications above about 500Hz. The capacitive load is within specification above 200Hz. The lower frequency limits were a compromise, as I wanted reactive loads that would resemble real-world loudspeakers, and I wanted to limit the number and size of the components needed.

As an amplifier designer I can attest that reactive loads usually have their worst influences above 1kHz. The total harmonic distortion (THD) introduced by the reactive loads measures less than 0.01%. It's probably much less, but

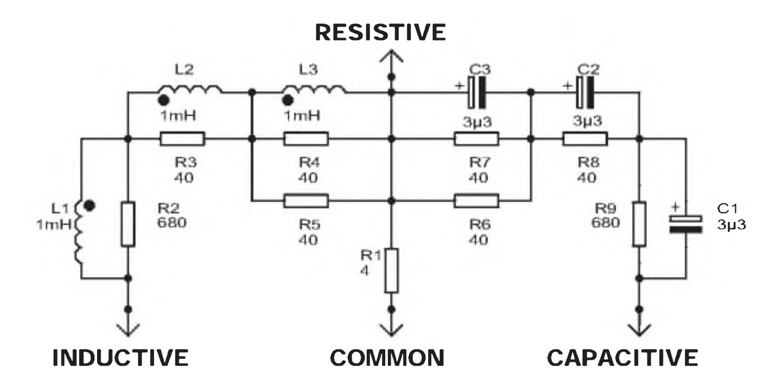

FIGURE 1: Schematic of reactive load.

this is the limit of my present measurements.

POWER RATINGS

The power dissipation of this load box is limited by the 4Ω, 100W resistor. For higher power you need a more robust 4Ω resistor, and for power levels above 200W I recommend more robust inductors, as well. I selected the capacitors and other resistors in the parts list to tolerate an amplifier output of at least 40V RMS, or a nominal 400W.

Heatsinks for the 4Ω resistor may be required for safety reasons, as my calculations show that even at 100W this design will become hot. I mounted the 4Ω resistor on top of the box to improve the cooling due to airflow. Both of the reactive loads have lower power dissipation than the resistive load.

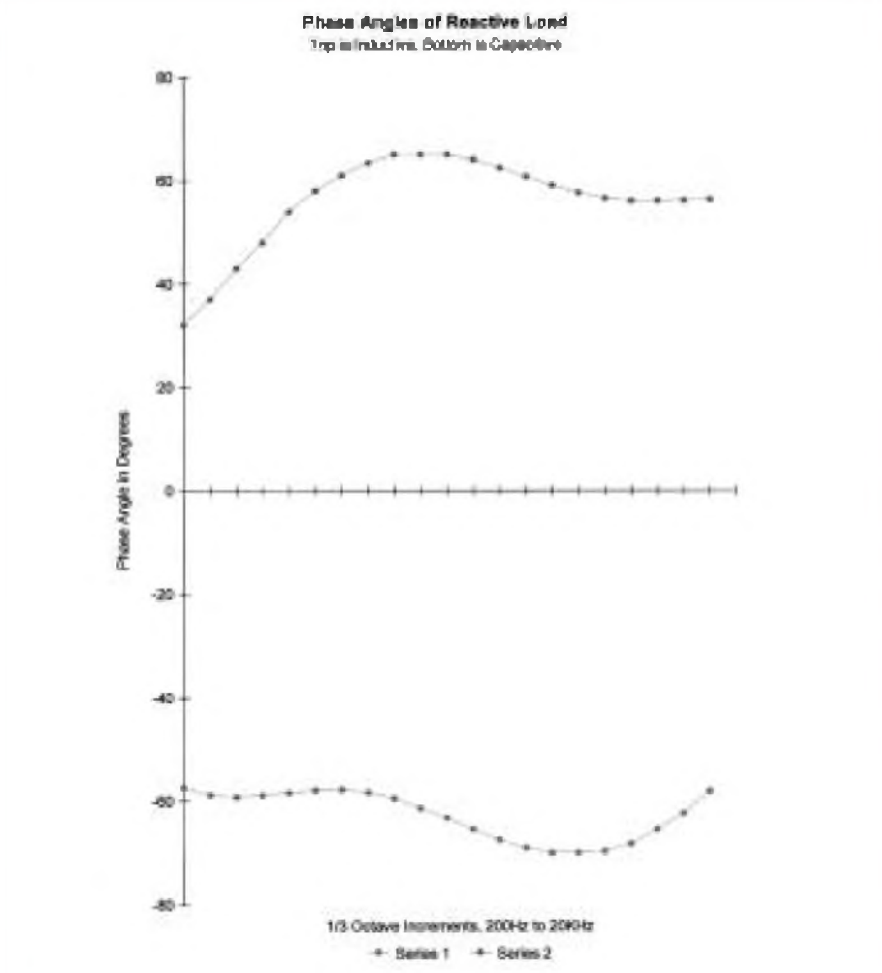

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FIGURE 2: Phase angles of reactive load.

TABLE 1
REACTIVE LOAD PARTS LIST FOR SINGLE CHANNEL AT 100W

		COST
CAPACITORS:		
C1–C3	3.3µF/250V, 5%, 027-420, 3 @ $1.77	$5.31
INDUCTORS:		
L1–L3	1mH, 2%, 300W, 266-826, 3 @ $5	$15
RESISTORS:		
R1	4Ω, non-inductive, 100W, 019-015, 1 @ $11.50	$11.50
R2, R9	680Ω, 10%, 5W, 2 @ $0.19	$0.38
R3–R8	40Ω, 10%, 20W, 6 @ $0.55	$3.30
HARDWARE:		
CONNECTORS:		
B1–B3	Red binding post (banana), 090-485, 3 @ $0.95	$2.85
B4	Black binding post (banana), 090-480, 1 @ $0.95	$0.95
Chassis	Radio Shack project box, 8 × 6 × 3″, 270-1809, 1 @ $6.99	$6.99
	Total:	$46.28

Components from Parts Express, unless otherwise noted

DESIGN OPTIONS

What about 8Ω loads, or 2Ω loads, or whatever? The general rule is that if the load impedance is doubled, then all the inductors and resistors of *Fig. 1* are doubled, and all the capacitors are halved. Of course, the "versa is vice."

CONSTRUCTION

I built my unit in a Radio Shack project box. All of the other parts are from Parts Express (*Table 1*). I mounted the 4Ω resistor to the outside of the metal top using 8/32 hardware. The other parts are mounted to the inside of the metal top using silicone rubber glue.

I glued the inductors so that they are perpendicular to the aluminum top—close to, and parallel to, the sides of the plastic enclosure. This gives good separation of the inductors and reduces eddy currents in the top. I also glued the 40Ω, 20W resistors to the inside of the top cover.

There is no printed circuit board, as I made the connections by wrapping the component leads around each other and applied lots of solder. I drilled several ¼″ ventilation holes in the plastic enclosure.

RESULTS

I measured the total harmonic distortion (THD)–at 2.83V RMS–of one of my newer amplifier designs using this load box. As compared to the resistive load, the THD increased moderately for the inductive load, and just about doubled for the capacitive load. The results for the intermodulation tests that Mr. Colin recommends might be different.

CONCLUSION

There is no doubt that reactive loads are harder for an amplifier to handle than resistive loads, and so it is instructive to test power amplifiers with this reactive load box. This design is simple, the cost moderate, the size is reasonable, and it's easy to use. ❖

Care and Maintenance of DuKane Ionovac Tweeters

Discover how these unique speakers operate . . . and how you can bring them back to life. **By Daniel Schoo**

The DuKane Ionovac (*Photo 1*) is distinctive among speakers in that it operates on a principle very different from the usual mechanical transducer with moving parts. In the Ionovac there are no parts of the speaker that move. The sound is produced by the interaction of air with high temperature plasma contained inside an open-ended quartz tube.

HOW IT WORKS

The system is based on a radio frequency oscillator operating at about 27MHz. The audio signal applied to the oscillator modulates the amplitude of this radio frequency signal. The RF signal from the oscillator is greatly increased in voltage by a coil very similar to a Tesla coil. The output of this coil connects to a small pointed electrode inserted into the quartz tube. Energy from the high-voltage RF signal radiates from a sharp point on the end of the electrode and forms a plasma inside a small chamber in the quartz tube.

As the amplitude of the audio signal increases and decreases, so does the amplitude of the RF energy. This, in turn, causes the size of the plasma field to increase and decrease in direct proportion to the audio. As it does, it compresses and rarefies the air in the chamber causing compression waves. These waves exit the end of the tube and are coupled to the inlet of a horn. The waves travel down the horn and exit into the room as sound.

The application of this simple concept is closer to a perfect speaker than just about any other means. With no moving parts, mechanical inertia and resonances are minimized.

While it is a very good speaker, it is far from perfect. Nonlinearities in the modulation of the 27MHz RF and colorations added by the transformer that couples the audio into the oscillator, called the modulation transformer, alter the frequency response and add distortion. These effects are relatively small compared to the typical voice coil or electrostatic type of speaker with the inherent mechanical drawbacks each has.

Also, because of physical limitations, the lower end of the frequency response is limited to about 1.5 to 2kHz. Frequency response above this extends well into the ultrasonic range and is limited mostly by the modulation transformer.

There are two sections to the Ionovac speaker connected together by a four-conductor cable and ground strap: the power supply (*Photo 2*) and the oscillator/horn assembly. The power supply contains the components that convert the 120V AC power to the DC supply voltages necessary to operate the oscillator. The power supply is a voltage doubler type with no power transformer. The main supply is 310V DC for the plate of the oscillator tube and 150V DC for the screen grid.

Two other parts complete the power supply. A small 120V to 6.3V filament transformer supplies power to operate the heater in the oscillator tube. A modulation transformer couples the low impedance signal coming from the audio amplifier into the screen grid of the oscillator tube. The modulation transformer also provides ground isolation

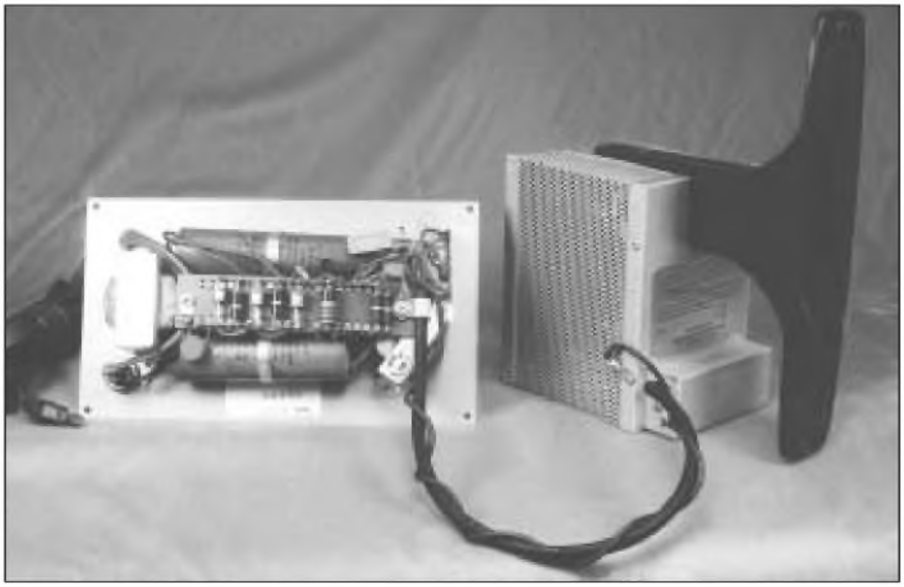

PHOTO 1: Overall view of the complete speaker system.

ABOUT THE AUTHOR

Daniel Schoo's interest in antique radios and related equipment began at a young age. He collected radios and learned to repair them growing up in the mid 1950s and 1960s. After graduating from college in 1973 with a Bachelor's degree in Electronics Engineering Technology, he accepted a position at the Fermi National Accelerator Laboratory, a national laboratory for research in high-energy physics, where he continues to work. His interests cover many facets of electronics and antique radio technology.

of the 120V AC power line and blocks the 150V DC screen bias voltage from appearing at the audio input terminals of the Ionovac. These components are all mounted on an open-face metal plate and tied together using a set of terminal strips.

The oscillator/horn assembly contains the oscillator tube and related components, the high voltage coil, the electrode/quartz cell (*Photos 3* and *4*), and the horn, all mounted in a shielded aluminum box (*Photo 5*).

REGULAR MAINTENANCE

The biggest problem in using Ionovac speakers is maintenance. The construction of the Ionovac is much more complex than a magnetic or electrostatic speaker. With consumable components involved, more can go wrong.

When a critical component fails, the speaker is useless until you replace the part. The electrode is consumed during operation and the quartz tube becomes encrusted with dirt and oxidation products of contaminants in the air and electrode residue. A small screen inside the inlet of the horn also becomes clogged with dust and oxidized byproducts. Eventually the tip of the electrode burns away, and the speaker functions poorly or ceases to function. The vacuum tube becomes weak and other components fail.

DuKane guaranteed a minimum lifetime of 1200 hours for the cell. When it did need replacement, DuKane sold a kit containing an electrode and a quartz tube called a "cell replacement kit" part number 438-37, which you could buy for a few dollars. The kit included instructions on how to remove the old cell and install the new one.

When the electrode went bad and the quartz became fouled, installing a cell kit would restore the operation. This assembly is sometimes called a "crystal," but that term is not used in any of the DuKane documentation. Many years ago DuKane discontinued supplying Ionovac parts, which are now very scarce. Fortunately, very few of the components prone to failure are absolutely irreplaceable.

First and foremost in operating failures is the electrode. When you first turn on the speaker, the oscillator tube takes some time to warm up before it starts. When the oscillator starts, you hear a slight click and see a blue-violet glow down inside the throat of the horn (*Photo 6*). The glow initially is unstable and sometimes takes a minute or so to fill in and settle down as it reaches operating temperature.

After it settles down, the glow should be perfectly round and even in intensity over the entire end of the tip of the electrode except for a small brighter spot in the exact center. Some electrodes on startup emit a buzzing or squealing sound as the tip heats up and the unstable plasma pulsates. DuKane called this effect "singing," which is a normal part of startup.

With a new electrode installed, the plasma can be uneven and produce these effects as it breaks in but should clear up within about the first 15 minutes of operation. If the plasma does not fill in evenly or continues to make

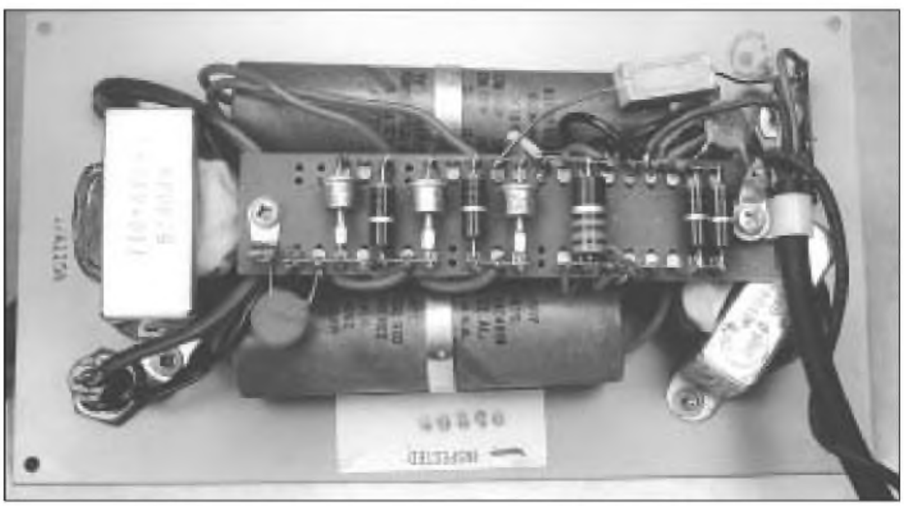

PHOTO 2: The power supply is a simple circuit. The transformer on the left is the filament transformer and the one on the right is the modulation transformer.

any sounds of its own, you should remove the electrode and quartz and examine them for damage, wear, and fouling. If the electrode is bad, you must replace it. Other than individual private stocks, there are no sources to buy new electrodes. This does not mean that you cannot replace them; they can be specially made if you have the production documentation.

You should take special care when removing an electrode. Never grasp the lead wire and pull on it to retract the ceramic retaining bar. This will overstress the bar and possibly snap it in half.

The retaining bar is made of a machinable glass-bonded mica ceramic called Supramica™. I have made replacement retainers out of Macor™ machinable glass ceramic that work equally well. Machinable ceramics are relatively expensive, and the fabrication of replacement retainers as well as electrodes is beyond the scope of this article.

Grasp the two ends of the bar between the thumb and forefinger of one hand. Hook the connecting wire with your middle finger and apply a slight tension to it so that the cup remains seated in the retaining bar as you retract it. Pull straight back on the bar until the connecting cup just clears the end of the electrode (*Photo 7*). If the cell starts to come out with the connecting cup, push it back into place with a small wood stick and hold it until the cup is clear. Place the bar below the cell and slowly allow it to return to a forward position underneath. Remove the electrode and quartz for service. Replace the electrode and quartz and place the ceramic retaining bar back in position (*Photo 8*) in reverse order of removal.

ELECTRODE LORE

DuKane has long since discarded its original documents on the electrode. Even though the DuKane documents are gone, much is known about the electrode material and dimensions. The DuKane electrode (*Photo 9*) was made of a metal alloy very similar in composition to type 416 stainless steel. The manufacturer started with a rod of material drawn down to .162″ in diameter and turned down this raw stock to form the tip and shank ends and then cut it to length. After machining and finishing, the electrode was heat-treated to impart hardness.

Within limits the specific material is not as important as the dimensions and heat conductivity. Alloys that are hard are better than softer ones. Those having a good resistance to the effects of heat and corrosion along with a moderate heat conductivity make good choices for electrode service. Aluminum and other soft metals are totally unsuitable

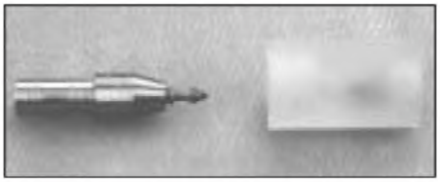

PHOTO 3: Close-up view of an electrode and cell.

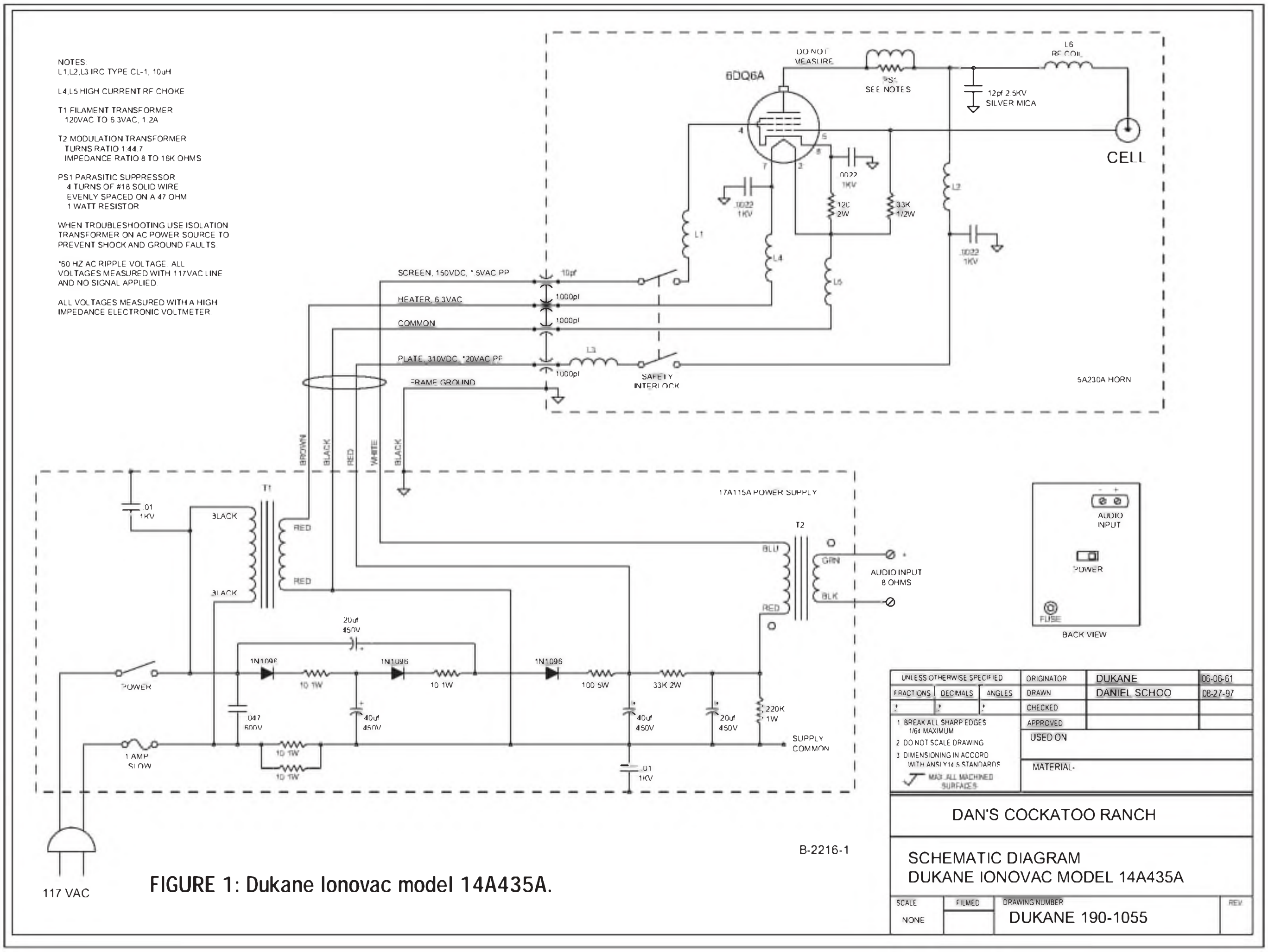

FIGURE 1: Dukane Ionovac model 14A435A.

for electrode material.

The inventor's original plasma speaker designs used platinum electrodes. Platinum is a good material but totally unsuitable for mass production because of the prohibitively high cost. Replacement electrodes have been made from various metals and alloys. The material used determines the operating temperature, the durability, and lifetime of the electrode.

PHOTO 4: Electrode inserted in a cell. The second electrode is for reference to illustrate the relative position of the parts of the electrode inside the cell.

Several of the stainless steel and nickel alloys are good candidates for electrodes. Type 304 is a commonly available stainless alloy that is relatively easy to machine and has a moderate life comparable to that of the factory originals. High-temperature space-age alloys such as Inconel™ would also probably work very well.

Tungsten and tungsten alloys make outstanding electrode materials. Tungsten is extremely hard and resistant to heat. It can last almost indefinitely but is very difficult to shape due to its hardness.

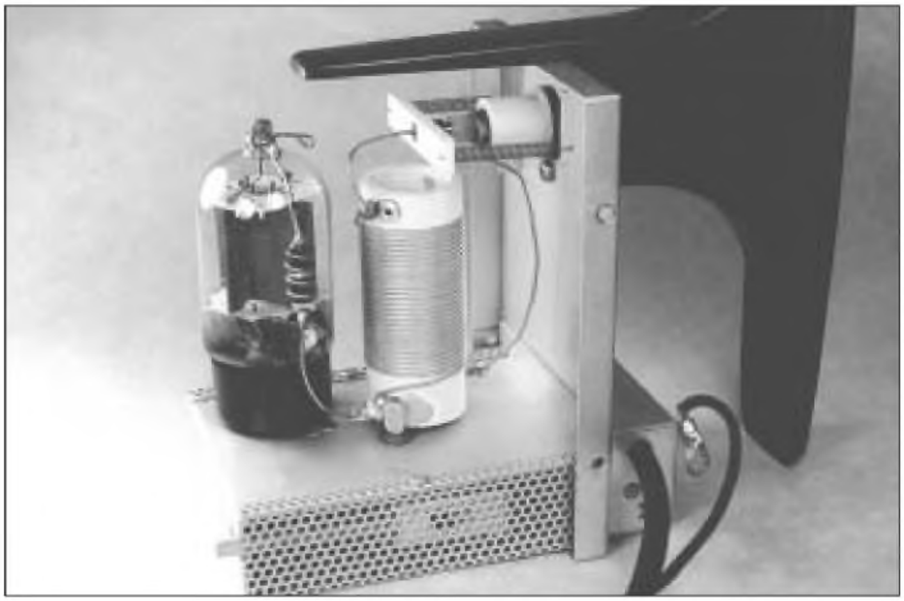

PHOTO 5: The speaker section with the protective cover removed. Note the small tab in the lower left corner of the chassis. This is the actuator for the interlock switch that disconnects the supply voltages when the cover is removed.

Another disadvantage of tungsten is its heat conductivity. High tip temperatures are important for proper operation. Tungsten has superior heat conductivity, and because of this, tungsten electrodes can take a little more time to reach operating temperature than nickel or chromium alloy electrodes.

Claims that the surface must be absolutely smooth and flawless are not accurate. I have deliberately made electrodes with a slightly rough surface, and they worked as well in tests as highly polished samples. The proof of this is evidenced by the fact that no matter what the finish is when first installed, the electrodes soon become pitted and burned in service, and they continue to function satisfactorily.

QUARTZ CLEANING

Another cause of failure is fouling of the quartz tube. When replacement parts were available, you could discard and replace the electrode and quartz tube with a new set. Today, because this is nearly impossible, you must reuse the quartz.

Fortunately, the quartz is relatively sturdy and will outlast several electrodes. Dirt and deposits from the high plasma temperatures build up on the inside of the plasma chamber and can obstruct the opening. As long as the quartz tube is not broken, cracked, or has no serious pitting, you can clean it and place it back in service. If it is damaged beyond use, there is no alternative at this time to hunting down an old stock replacement at great difficulty and, no doubt, great expense. I have tried several powerful solvents and acids to remove the crud inside the quartz tube, but have found no chemical method that will remove the deposits effectively.

The best method turns out to be the simplest. Gently scraping the inner surface of the quartz tube will remove nearly all of the built-up deposits. A scraper having a very hard surface works best because it will not rub off onto the rough surface of the quartz.

I have used the blunt end of a number 68 solid carbide drill bit to clean off the surface. A very gentle scraping, one stroke at a time, in and out all the way around the inside surface removes nearly all of the deposits. Do not gouge or attempt to use force, which can damage the surface of the quartz. Better to leave some crud on the quartz than to damage it. Perfect cleanliness is not necessary, but remove as much of the deposit as you can.

After the scraping, gently swab out with a cotton swab soaked with a mild detergent solution and rinse in distilled water to remove the remaining loose dirt. Following a thorough air drying, the quartz tube is ready to use. Never apply force to push anything down through the opening of the quartz, which can be easily cracked and broken.

The screen inside the entrance of the

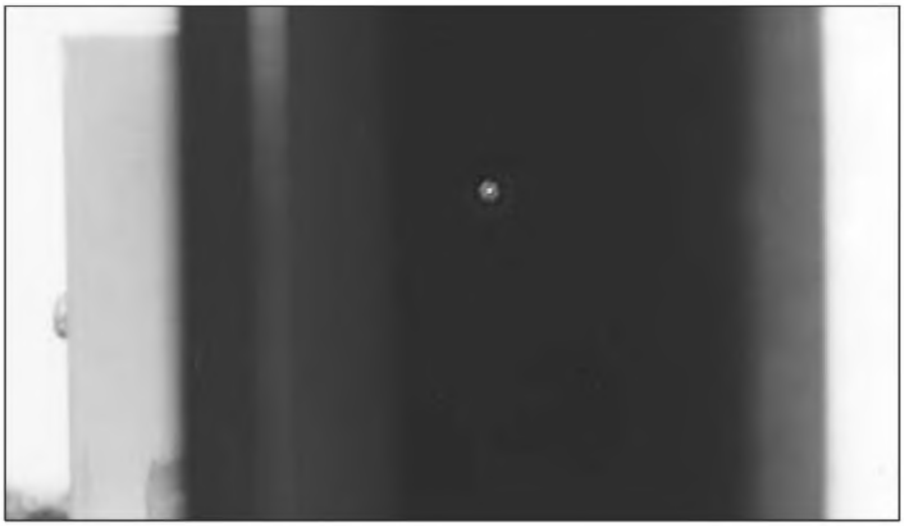

PHOTO 6: A normal plasma pattern. Note the even circular glow with a brighter point in the center. The crosshatch distortion is due to the mesh screen located just in front of the cell in the throat of the horn.

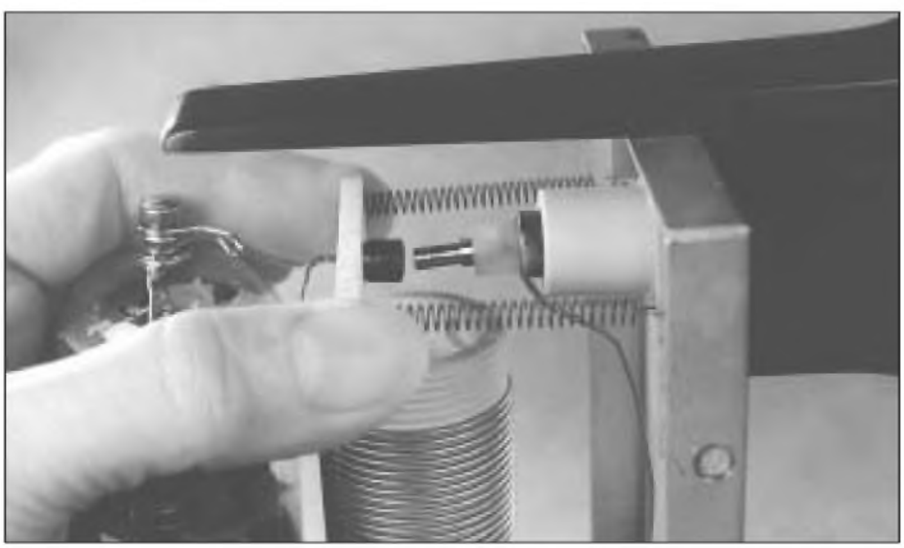

PHOTO 7: The correct way to hold the ceramic bar for removal of the cell components.

horn also collects dirt, which can obstruct the opening. Remove the cell from the horn, then give the horn a gentle blast of air down the throat from a canned air duster. This will frequently remove the accumulated dirt. If this does not remove all of it, very gently rub the screen with a long cotton swab to help loosen the dirt. Be careful not to apply much pressure to the screen, which is very thin and can be damaged easily by too much force.

ELECTRODE CARE

When an electrode becomes worn with use, it can cause the plasma to become unstable or irregular. The end of the tip becomes burned and pitted, wearing back during normal operation. Difficult or incomplete ignition, buzzing, squealing, or snapping noises can result from this. New electrodes tend to do this before they are broken in, but this should clear in a short time. You can sometimes clean used electrodes that have not been completely consumed and put them back into service.

Grasping the tip in a folded-over piece of medium-grade Scotchbrite™ and rotating it is a good way to burnish and remove deposits. If that isn't sufficient, gently clean the tip with 600-grit sandpaper to help restore proper function. Place the paper on a hard, flat surface and gently rub the tip of the electrode forward and backward on the paper at an angle that puts most or all of the flat side of the tip against the paper. Rotate the shank of the electrode between your thumb and forefinger as you rub it on the paper so that you get even coverage. Just a few strokes are enough.

Removal of large amounts of material or sanding out deeper pits will shorten what life, if any, is left. If the end is burned off flat, wipe the paper a few times across the end to clean it, too. The idea is not to remove much base material but only some of the crust that forms and smooth out the larger deformations.

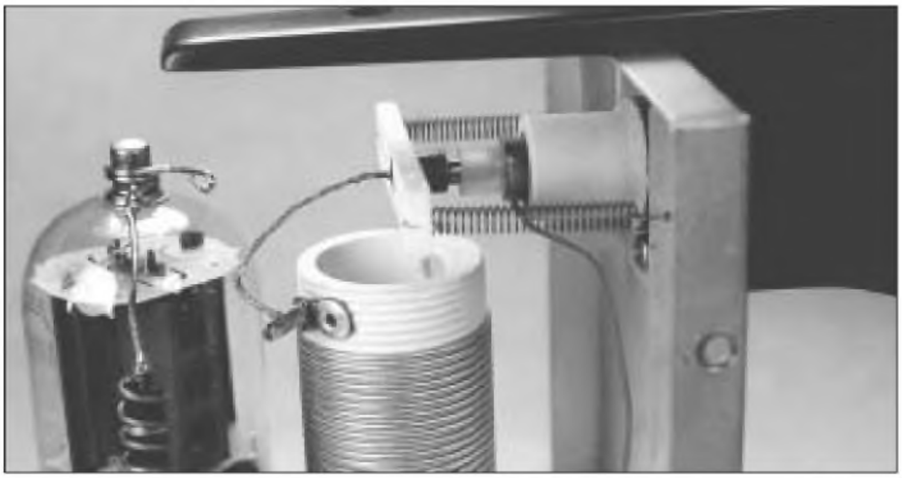

PHOTO 8: Close-up view of the cell mounting assembly. Note the electrode and quartz cell being held in place by the spring-loaded ceramic retaining bar. The bar holds a metal cup pressed against the back end of the electrode. A flexible braided wire attached to the cup connects it to the high voltage coil. Surrounding the cell is a metal sleeve isolated from ground. A wire attached to the sleeve connects it to the oscillator circuit.

COMPONENT FAILURES

Electrical failures in the associated electronics are common. I will describe several types of failures that happen and what you can do to repair them.

Oscillator Tube Plate Cap

Every single Ionovac I have seen has had the plate cap clip on the oscillator tube badly corroded and overheated. The original construction used a plate cap clip composed of two half sections

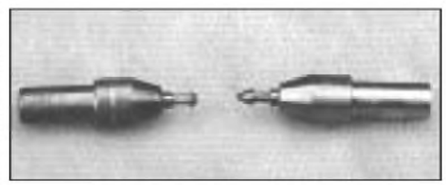

PHOTO 9: Comparison between a new electrode on the right and a used electrode on the left. Note the difference in the length and the sharpness of the points.

held together by a rivet. The lead wire was put through the hole in the rivet and soldered.

After years of use the rivet corrodes and loses electrical connection with the contact parts. The rivet heats up and becomes progressively worse. Replacement of the plate cap connector is mandatory on all units. I have used the spiral spring clip type with good results.

Power Supply Cable

The connecting cable between the power supply and the oscillator/horn is insulated with a rubber jacket. With age the rubber becomes hard and cracked. With any movement the rubber breaks off the wires exposing them. Replacement with a new four-conductor #18 cable will solve this problem. The original cable is Belden type 8454 and is still available.

You must take great care when unsoldering the old wire and resoldering the new wire to the feedthrough capacitors in the oscillator/horn assembly. Overheating can melt the solder in the feedthrough causing the center lead to slip out of position. Excessive stress on the capacitor lead can break the insulating sleeve.

Feedthrough Capacitors

Feedthrough capacitors can become broken or arced over if the speaker is subjected to large overloads. I repaired one speaker—part of a DuKane column system—that had the woofer blown out by seriously overdriving it. The associated Ionovac tweeter had an arced-over feedthrough capacitor in the oscillator/horn that carried the screen bias and audio signal. It had blackened areas on it where the arcing had burned the metallization.

To prevent damage to the speaker, the absolute maximum audio input voltage should never exceed 2.16V RMS sine wave or 3V peak. Replacement is the only repair for an arced feedthrough. There are two values of feedthrough capacitors—10pF for the oscillator screen grid and 1000pF for the two heater leads and the plate supply. Replacement capacitors are still available from manufacturers on special order.

Modulation Transformer

Modulation transformers can also become damaged by the stress of being overdriven. They can develop turn-to-turn shorts in the primary or secondary. This will cause the speaker to work at greatly lower volume levels, add distortion, or fail to operate at all. In one speaker I worked on, the modulation transformer had developed a primary to secondary short. This allowed a current path from the power supply back to earth ground through the audio amplifier.

The speaker first blew its fuse. The owner replaced the fuse with one of a substantially higher rating. This, of course, prevented the fuse from blowing again but resulted in all of the DC power supply components and modulation transformer going up in smoke! When I first examined it, I found the modulation transformer and all of the power supply resistors were badly charred.

Replacement of the modulation transformer is not too difficult. Any good-quality audio transformer of a comparable physical size and impedance ratio will work. If only one speaker in a stereo pair needs repair, it is advisable to also replace the transformer in the other side so that any differences between the old transformer and the new one will be matched in the other channel.

When replacing the transformer, be sure to maintain the original phase polarity, which will guarantee that the speaker will match the phase of its mate and any other speakers in the system. I have marked the schematic diagram (*Fig. 1*) with the phase of the original modulation transformer to use as a guide for replacement.

Power Supply

The power supply is fairly reliable. Solid-state rectifiers have a very long service life. Only an overload is likely to cause a problem.

At this time all of the Ionovacs are at least 40 years old. The electrolytic capacitors are the most likely components to fail because of advancing age. To date I have not had to replace any electrolytics in an Ionovac, but I'm sure there are some that need it. There are two dual section electrolytics used in the Ionovac. If they need replacement, use four single types having the same values as the old ones.

The other components, such as rectifier diodes and resistors, are all commonly available parts. The original type diodes are no longer available, but modern ones can replace them. A set of 1N4005 silicon diodes will work well in this application.

Filament Transformer

Filament transformers rarely fail. If a good tube fails to light, check the tube socket for bad connections. Measure the filament circuit with an ohmmeter and verify that the heater RF chokes are not open. If the filament transformer has gone bad, you can easily replace it with any transformer of a comparable physical size and rating.

WORDS OF CAUTION

The voltages used in this device can be dangerous and possibly lethal if not handled correctly. Certain components such as the cell and the 6DQ6A tube become extremely hot during operation, so you must allow them to cool before doing any work. Remove all power completely from the Ionovac by unplugging it.

You should never bypass the interlock switch on the oscillator cage. Dangerous voltages are inside and present a serious shock and electrocution hazard. If you are not familiar with electronic devices, do not attempt to do the work yourself. Get help from a qualified technician.

Oscillator Tube

A weak tube will fail to oscillate or give only poor performance. If the cell is in good condition and the power supply voltages are normal, then you can suspect a bad tube. Substitution with a known good tube is the best test for a bad one.

Replacing the tube easily solves tube failure. The original tube is an RCA type 6DQ6A, which is not in demand for audio work and was primarily used as a horizontal output amplifier in black and white television sets. You can purchase them for no more than a few dollars from a number of tube suppliers.

REPLACEMENT PARTS

Feedthrough capacitors:
Tusonix type 357-001-X5U 100M, 10pF, screen grid
Tusonix type 357-001-X5U 102M, 1000pF, heater and plate

DuKane Ionovac modulation transformer:
Turns ratio 1:44.7
Primary impedance 8Ω
Secondary impedance 16,000Ω
Primary DC resistance .2Ω
Secondary DC resistance 519Ω
Frequency response: 30Hz–20,000Hz, +1, –0.5dB
Power rating: 5W
Suggested substitutes:
Thordarson/Meissner 24S74, 70.7V line to voice coil, using the .310W tap
Stancor A-8105, 70.7V line to voice coil, using the .310W tap

The closest substitute available for the DuKane modulation transformer is a 70.7V line to voice-coil transformer. Use the 8Ω speaker winding as the audio input and the .310W tap on the 70.7V line winding as the output to the oscillator screen. Except for slight variations in frequency response, the two listed transformers are an exact match for the original.

Filament transformer:
Primary 120V AC
Secondary 6.3V AC at 1.2A
Magnatek/Triad type F-14X
Stancor P-6134 (center tap is unused)

Tube source:

Antique Electronic Supply
6221 S. Maple Ave.
Tempe, AZ 85283
(602) 820-5411

Oscillator Components

Overdriving the speaker can damage components in the oscillator circuit. A failed tube can also cause trouble. These parts are readily available, so you can easily replace them. Overheating because of the bad plate cap clip can also damage plate resistors.

IMPORTANT SPECIFICATIONS*

Sound output:
.75V RMS continuous sine-wave input = 95dB at 1.5′ on horn center axis

Frequency response:
±3dB from 3500Hz–30kHz

Operating input impedance:
At 200Hz = 5.1Ω
From 1kHz–20kHz = 8Ω +0, –0.8Ω
At 30kHz = 6.30Ω

Non-operating input impedance:
At 200Hz = 8.6Ω
At 5kHz = 182.6Ω
At 20kHz = 43.7Ω

Input voltage into 8Ω:
0.75V RMS sine wave/1.06V peak = recommended for maximum undistorted output
1.50V RMS sine wave/2.12V peak = 50% modulation
1.75V RMS sine wave/2.47V peak = maximum input without parasitic oscillation
2.16V RMS sine wave/3.05V peak = 100% modulation, absolute maximum input voltage

*Ratings given are for a typical speaker with a new electrode. Your speakers may vary due to the individual characteristics of the oscillator tube and/or cell condition.

The capacitors in the oscillator section are common types except the 12pF 2500V silver mica. Fortunately, these are very reliable and seldom fail. I have seen one become intermittent, which caused the oscillator to start only once in a while. Silver mica capacitors at this capacity and voltage are very difficult to find. Since the measured voltage across this capacitor never exceeds 850V, I substituted one rated at 12pF at 1000V and had no problems.

The RF chokes in the oscillator are very unlikely to fail. If they do, you can replace them with modern parts that are designed for RF suppression and have about the same inductance. The values are not critical since the chokes are used only to suppress the RF from getting out of the oscillator cage and radiating into the air from the interconnecting power supply cables. ❖

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Showcase

An Italian THOR

I'd like to share with *audioXpress* readers my piece of work devoted to restyling THOR towers (*Photo 1*). The Joe D'Appolito project has been described in the *audioXpress* May 2002 issue, and a detailed, step-by-step do-it-yourself construction description has been published by Edward T. Dell, Jr. (Sept. 2002).

The following is my version of this project which departs from previous

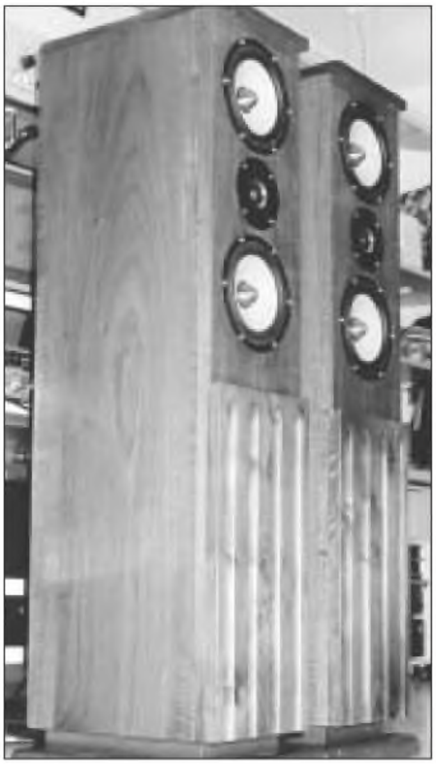

PHOTO 1: Overall towers view. You can see the provision made to accept the grille frame once ready. I'm thinking of using thin material similar to ladies' elastic stockings.

ABOUT THE AUTHOR

Atto Rinaldo is an active retiree after 32 years at IBM, where he experienced various job responsibilities in the US and other parts of the world. He received a radio techniques diploma 45 years ago. Vacuum tubes were the technology used at that time, and he has loved them since. Transistors were just born; he had never heard about integrated circuits. Joining IBM in the late '60s, he has seen vacuum tube computers phasing out just about that time. When transistors and ICs took over, he said to himself that one day he would play with tubes again. He did.

PHOTO 2: A view of the cosmetic panel. The grooves give the tower a "slim" look.

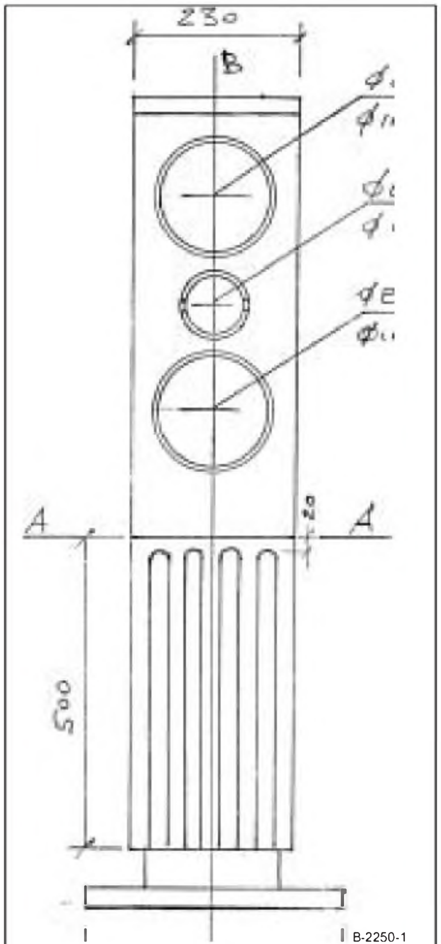

FIGURE 1: This is the original sketch that was selected as "the best" after debating different solutions with a couple of friends.

PHOTO 3: My friend Siro at work while refining a speaker's hole. MDF dust was his worst enemy.

ones in the area of external design. As it turned out, I was so delighted by what I had done that I wanted to share my experience with *audio-Xpress* readers.

While maintaining the exact specifications of all internal measurements—in order not to alter the quality/performance—I wanted to give this project a better look that its superb sound quality deserves. I thought that the following could be improved over previous projects/prototypes:

1. The bulky base. It needed restyling. Removing it would have given the tower a "bare" look and force you to install crossovers somewhere else.
2. The front view was too "dull." It required some work to make it more attractive.
3. Missing grille. I believed it needed one in order to protect the speakers from accidental touching.

With that in mind, I prepared a few sketches, and, with the help of some friends, who looked at the different options, I decided to proceed with the version depicted in *Fig. 1*.

The changes include the following:

PHOTO 4: "T-nuts" used to reliably install the speakers.

- The base has been restyled.
- The addition of an "aesthetic" panel on the bottom half of the tower (*Photo 2*).
- Provisions made to allow installation of a grille.

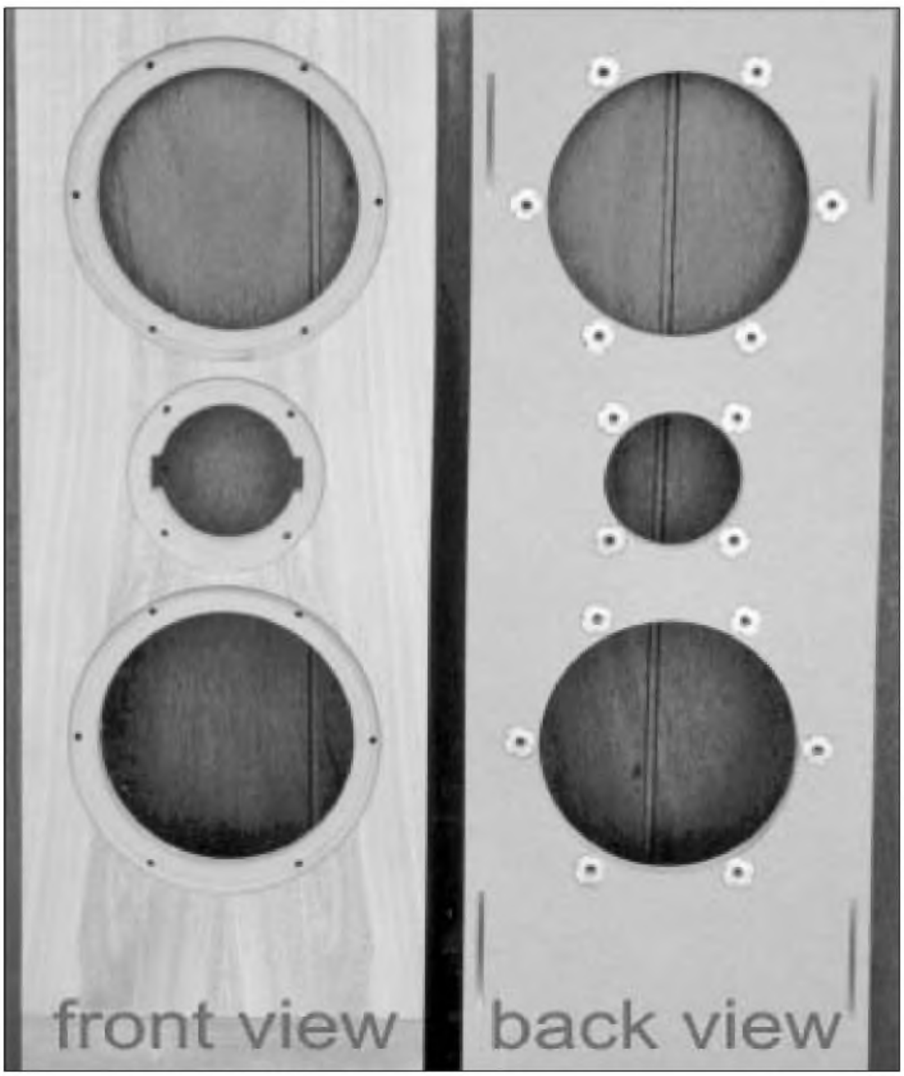

PHOTO 5: "T-nuts" are solidly mounted on the back side of the front panel.

Drawing a sketch was the easiest part, because building a high-quality THOR TL is not an easy task at all. With the help of my friend Siro (*Photo 3*), a professional joiner, I had most of the work done in his highly qualified joinery. He performed each step with extreme ease; however, he complained about the use of MDF, which, in spite of his dust collector system, seemed to produce dust all over. The sequence of photos, better than my words, give you an idea of the process and changes.

I would like to comment on the solution I've taken to prepare the holes to fix the speakers.

Since MDF isn't any good at retaining screws—especially if you must remove them a few times—I've implemented the solution of inserting, on the back side of the front panel, a number of "T-nuts" (*Photos 4* and *5*), which gives you the opportunity to remove the screws as many times as you want.

All major parts were assembled using "biscuits" and waterproof glue, as required.

The base contains the crossover (*Photos 6* and *7*) mounted on an aluminum plate which is screwed to the

PHOTO 6: A view of the base. You can see the wires coming through the panel before sealing the hole with silicon glue. Also visible are the golden speaker's terminals.

PHOTO 7: Crossover view before installing onto the base.

I congratulate reader Atto Rinaldo on a beautiful realization of Joe D'Appolito's THOR design using the quite impressive SEAS drivers. His base is certainly an improvement in appearance. The changes to add a grille have some hazards, however.

As Joe has often observed in his many reviews, *any* grille will change the character of the sound. Whether this is for the better or worse is not clear. It could add some amount of diffraction to the sound character both by the top and lower offsets for the grille, as well as the inner edges of the grille frame. His outer edges of the front panel also do not appear to be rounded, another diffraction issue.

The decorative fluting in the lower front section is elegant, but is another variable whose sonic effects are not known. My much plainer design was chosen for purely acoustic reasons, and, since I live alone, I am the only one it needs to please. My visitors have, so far, been so awed by the sound, and they have not commented on the lack of front panel elegance. I was also anxious to strictly follow the designer's prototype.

The Madisound version of the THOR cabinetry also adds a grille to the box. Reader Rinaldo's crossover differs from mine in that bi-wiring is not possible in his. My tweeters and bass drivers are driven by separate amplifiers, theoretically reducing IM distortion.

I fully intended to use threaded brass inserts for mounting the THOR drivers with machine screws, but had not as yet devised a way to position them accurately. Readers may notice that in my Froy project in the March 2003 issue, I finally managed to successfully use the brass insert/machine screw combination to mount the SEAS hardware. Reader Rinaldo's solution is excellent and widely used by most speaker builders.**—E.T.D.**

bottom of the base itself. The base is attached to the tower with four screws to the bottom side of the "corners," once the filling material has been properly put in place. To allow the screws to firmly hold the base in place, the corners were made of 1.5″ marine multi-layer wood instead of MDF.

Of course, the varnishing on the veneering took place just after the final assembly. I selected a walnut color to match the furniture of the house. Installing the speakers was the last step.

Finally, it was time to hook them up to my 90 lb 8 × EL34 25W power amplifier driven by a PST 200 preamp—both self-made, vacuum tube tech—and listen to the splendid voice of Amanda McBrown on her LP direct disc recording. *audioXpress* experts have described the exceptional sound quality of THOR TL. They were right on. This is an outstanding piece of audio equipment. Audiophiles who do not have them or have not listened to them, do not know what they're missing.

Thanks, Joe, this is really a great project (*Photos 8* and *9*).

Atto Rinaldo
Tambre, Italy

PHOTO 8: After all that work, an homage to Joe D'Appolito, "thumbs-up."

PHOTO 9: Final version of THOR with grille. On the left, a black version for a friend who fell in love with the sound.

DAVID HAFLER

David Hafler, 84, who devoted his life to perfecting home high-fidelity sound components, died in Philadelphia on May 25 of complications of Parkinson's disease. A life-long resident of the Philadelphia area, Mr. Hafler also owned homes in Boca Raton, Fla., and London. Naturally, all of his homes were filled with music from quality sound systems he designed, said his daughter Joan Cole. He didn't watch television, she added.

Born and raised in West Philadelphia, Mr. Hafler graduated from West Philadelphia High School in 1936, and received his degree in mathematics from the University of Pennsylvania in 1940. His life changed after the Japanese invaded Pearl Harbor. Mr. Hafler quickly married his high school sweetheart, Gertrude Schwinger, and he enlisted in the Coast Guard, which made him an officer.

Both were very good moves. The couple remained inseparable until her death in 2001. During the war, while serving as communications specialist in the Caribbean, Mr. Hafler was exposed to the notion that sound could be reproduced faithfully. After the war, Mr. Hafler worked for A.J. Wood, a market-research firm in Philadelphia, until his love for music spurred him to design easy-to-assemble electronic sound equipment for consumers.

In 1950, Mr. Hafler founded Acrosound in Roxborough, which built and sold audio transformers. It was his next venture, Dynaco, which he founded in 1954 in West Philadelphia, that set the standard for home music systems. Dynaco manufactured and sold amplifiers as build-it-yourself kits.

Mr. Hafler sold Dynaco to Tyco in 1968 and served as an adviser until 1971. One year later, he founded another company, David Hafler Co., manufacturer of inexpensive kits and preassembled hi-fi gear. Mr. Hafler sold the firm in the early 1990s to Rockford Corp. of Tempe, Ariz. In 1984, he was named to the Audio Hall of Fame. One of his products, the classic Dynaco Mk. II 50W amplifier, was part of the media display in the Smithsonian's Museum of American History in Washington in the 1990s.

Another of Mr. Hafler's interests was collecting rare chess sets, which he was able to do because his business took him all over the world. Mr. Hafler's 240 beautiful and historic chess sets from across the globe were described by Sarah Coffin, specialist in chess collections and consultant to Sotheby's auction house in New York City, as "one of the finest collections of chess sets in the world." "The theme of my father's life was that if he couldn't be a top player, he could be a top manufacturer or collector," Cole said. "He loved music. But he couldn't play well enough to play professionally. So he manufactured the best sound equipment possible. He also loved to play chess, but he wasn't a world-class player. So he became a world-class collector."

In addition to his daughter Joan, Mr. Hafler is survived by daughter Diane Marinoff; a son, Eric; five grandchildren; and two great-grandsons.

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PHONO CARTRIDGES

My thanks to Ray Futrell for "doing the math" for the moving-magnet (MM) cartridges that most of us use to play LP records ("The LP Terminator," Jan '03 *aX*, p. 8). The most surprising result of Mr. Futrell's analysis is that following the manufacturer's recommendations in terminating the cartridge virtually guarantees sound that is shrill and unpleasant. This has been precisely my experience and explains why, for about 20 years, I have avoided moving-magnet cartridges altogether.

Because of its tendency to accentuate surface noise, the MM cartridge has been the largest contributor to the demise of the LP record. Mr. Futrell's research suggests that any MM cartridge (with the possible exception of the Grado and certain oddball cartridges from Stanton and GE) loaded with more than about 100pF of capacitance is going to have a peak in the high-frequency response that will grossly magnify this noise.

In practice, all installations will have more than 100pF worth of shunt capacitance. The lowest-capacitance interconnect cables I am aware of—solid-conductor steel-cored quadraphonic cables from the '70s—have a capacitance all by themselves of about 100pF. The question then arises, for whom were these cartridges designed?

As I recall, in the case of the Shure, the answer at the time was to manipulate the standard 47k termination resistance. Some authorities recommended dropping the input resistance by around 30–50% and adding shunt capacitance. This would produce a gradually rising hi-frequency response that could have been partially neutralized by the tone controls.

My first instinct was to question Mr. Futrell's results because they do not take into consideration the mechanical response of the cantilever, but I recalled a frequency-response measurement made in the '70s on my cartridge of the time, an ADC XLM, showing a 3dB peak around 19kHz. This suggests that the output, at least at the higher frequencies, is indeed dominated by electrical resonance.

This was far from an uncommon practice—millions of cassette decks resonated their playback heads in order to extend high-frequency response. The difference between the phono cartridge and the tape head is that the maximum frequency ever seen by a tape head is limited by its gap width; there is no such limit in a phono cartridge, and in fact, ultrasonic pulses are commonly generated by minute imperfections in the surface of the typical record. The cartridge, in response, rings like a Chinese gong, generating a pulse train every time it hits the tiniest piece of debris. This effectively stretches the width of the pulse, which is equivalent to lowering its frequency. I believe this is the mechanism responsible for accentuating the surface noise.

It is a mind-opening experience to play an LP record that has very high levels of surface noise with a cartridge employing a crystal or ceramic element. The quality of reproduction might not be very high, but the surface noise seems to vanish almost completely! Since the principal objection to the LP has been excessive surface noise, it appears that the venerable LP has been getting a bum rap.

To hell with frequency response! We need further study of the MM cartridge in the time, rather than the frequency, domain. This is largely unexplored ground. It may be possible to damp these resonances, or to raise them sufficiently so that they have negligible effect within the audio band by lowering the Q of the cartridge, fiddling with the load resistance, and/or further reducing the shunt capacitance of the typical installation. Perhaps some sort of magnetic rather than electrical damping would be more effective.

I have long used only moving-coil (MC) cartridges in my system. These cartridges have output well into the mid-ultrasonic range, but rarely cause serious problems because they are (or should be) heavily damped. The noise output is therefore directly proportional to the disturbance.

Resonance, in fact, is a dirty word to me. I find that even small amounts of ringing in my cartridge produces large amounts of glare, vagueness of image, loss of detail, and listener fatigue. Fortunately, ringing in moving-coil cartridges can be very effectively controlled by adjusting the load resistor.

In contrast to MM cartridges, the proper load resistance of MC cartridges has been very poorly specified, where it is specified at all. A case in point is the old Dynavector 10A. This high-output cartridge was designed to be a plug-in replacement for a MM unit, and so it was—as far as output voltage was concerned. The cartridge, when connected to the standard 47k resistance, had a nasty, spitty high-frequency response that could be completely tamed by dropping the load resistance to around 100Ω or so (I think). It became my favorite cartridge for years.

In general, the proper way to determine the load for an MC cartridge is to clip in a test resistor and listen to the results. The sound will become deeper, richer, and more refined as you lower the resistance until the image suddenly collapses. The optimum load is perhaps twice the value that causes the image to collapse. Not very scientific, but with only a little practice, you will be able to find the perfect load for any MC cartridge in a matter of minutes. Hope this helps.

Bob McIntyre
Toledo, Ohio

Raymond Futrell responds:

Thank you for the comments on the article. I was surprised after "doing the math" and did not believe the results at first. The calculated capacity values are what you would expect for an RF circuit, not an audio circuit. However, two other engineers couldn't find a mistake in the math, so that meant either Shure or the filter theory was wrong. The Leach preamp experiment clearly demonstrated the sonic benefits of low shunt capacity. Reduce the shunt capacity and hear the music.

Reducing the shunt capacity decreases the Q and increases the bandwidth, ω_c, *because*

$\omega_0 = 1/\sqrt{LC}$ is increased. Reducing the load resistor isn't the best way to lower the Q because it reduces both the Q and the bandwidth, ω_c. The rule of thumb is: reduce the capacity to lower Q, increase the resistance to raise Q.

Increasing the bandwidth also reduces pulse stretching, which is proportional to the rise time, which is related to the bandwidth by the formula $T_r = .7/B$. Narrower bandwidth means slower rise time, and a pulse is stretched by the addition of the rise and fall times. This may be why wide bandwidth MC cartridges are less affected by record surface imperfections.

There are several ways to reduce the shunt capacity. Minimize the wiring between the cartridge and the preamp input. Keep it short and simple. You can make interconnects out of high impedance (low capacity) coaxial cable. RG-62 has about 13pF/ft of capacity, so you can have a reasonable length of this cable.

A big source of shunt capacity is the Miller effect in the input stage of the preamp. It's reduced by the loop gain, which may be low at high frequencies. An input stage with low Miller effect, such as a cascode, is best. There's a simple capacity cancellation technique that you can use with a differential input stage to reduce Miller effect.

The electrical damping probably has no effect on mechanical resonance because of the weak magnetic coupling between the coils and stylus. Even so, it's still a good idea to damp the cartridge electrical resonant peak and eliminate that problem.

If most MC cartridges are highly overdamped electrically, it seems reasonable that increasing the terminating resistor for a Q of .5 to .7, which provides the flattest and widest frequency response, would also provide the best stereo image. The overdamped frequency response decreases monotonically and is never flat. Also, the lower the Q, the faster the response decreases. This means that a Q less than .5 peaks the low-frequency response, and a Q greater than .707 peaks the high-frequency response.

I had always noticed a grainy harshness on some LPs with excessive high-frequency content, especially on vocals. After reading the "LP Terminator" article by Raymond A. Futrell, I decided to run the numbers for my Grado Red to see what kind of load capacitance it required. I found that for a Q of .707 it needed about 100pF to terminate the 45mH inductance sourced by the cartridge.

Unfortunately, the measured capacitance on my phono patch cables is around 220pF. I didn't wish to change these custom-made cables that use twisted pair/braid shielded silver with Teflon dielectric (old Teledyne Thermodynamics instrumentation cable from Boeing Surplus), so I needed to do something else.

I found that lowering the load resistor from 47k to about 32k and running the numbers yields about 110pF load for a Q of .707. After installing the resistor I noticed a not-subtle improvement in the high frequencies. Gone was the grainy hash overlying vocals. Cymbals sounded more natural and transients were sharper, but not fatiguing like before.

In the past I had tried to remedy this by adding more capacitance! I can now see what this must have been doing to the signal by looking at the author's square-wave simulations. All that hash I was hearing was the ringing of the resonant circuit. Since there aren't many test LPs with square waves, I have no way to confirm this, but my ears tell me of great improvement! So, if you have no way of reducing the load capacitance, try lowering the load resistance, which worked for me.

Kevin Elliot
Elliot Studio Arts
Seattle, Wash.

Raymond A. Futrell responds:

Congratulations on solving your problem. With 45mH of cartridge inductance, even several hundred pF of shunt capacity would still result in a very high ω_0. In that case, you can reduce the load resistor to reduce the Q. The cartridge electrical frequency response will still exceed the audio bandwidth. With a high inductance cartridge you can't do that. The shunt capacity must be very low or the bandwidth will be less than 20kHz.

Don't reduce the load resistor below about ten times the cartridge resistance. The load resistance and cartridge resistance forms a voltage divider (attenuator), which reduces the signal-to-noise ratio. Lost signal-to-noise ratio cannot be restored by subsequent amplification.

Thank you for printing Raymond Futrell's article "The LP Terminator"! I have long been a proponent of reducing the capacitive loading on modern car-

tridges to a minimal value (see www.hagtech.com/loading.html). It's great to see such confirmation in a respected audio journal. Kudos to Raymond for helping to dispel this myth.

When designing my phono stages, I went to great lengths to keep the input capacitance as low as possible. The tube stages are under 40pF and the op amp stage is under 15pF. These values are far lower than normal and greatly improve treble clarity and transparency. Even cheap cartridges sound surprisingly good with no shrillness.

Jim Hagerman
Honolulu, Hawaii

Raymond A. Futrell responds:

Thank you for the comments. I was unaware of your very informative white paper on the subject of phonograph cartridge loading. Your simulations of various cartridge types clearly show why the proper load is important, even for the low inductance cartridges. With a low input capacity phono preamp such as yours, even a high inductance cartridge can be properly loaded for a maximally flat bandwidth over 20kHz.

The proper cartridge loading also makes cheap records sound surprisingly good. I have a budget label record of Baroque music containing lots of brass, percussion, and strings. I initially thought the record had gross distortion. After correcting the cartridge load capacity, the record sounded quite good. Evidently, the restricted bandwidth, frequency peaking, and ringing was causing most of the distortion.

MOSFETS AND BIPOLARS

I thank Doug Self for his reply to my letter (Jan. '03 *aX*, p. 61) in regard to bipolar vs MOSFET output stages. It looks as though we'll continue to disagree, with Doug remaining in the bipolar camp and me remaining in the MOSFET camp. We both agree that each has its advantages and disadvantages. I must take some exception to a couple of points Doug made.

Doug suggested that the Type II output stage can provide greater turn-off current to the bipolar output transistor at times of high current output. I disagree. A simplified version of a 100W version of the Type II stage is depicted in *Fig. 1*, where Vb1 and Vb2 model the usual bias circuit. One advantage of this stage over the more conventional stage is that the driver emitter followers operate in Class-A—they are never supposed to turn off.

The voltages shown are for the quiescent condition, with 1.5V across R1, setting the driver standing current at 10mA. The current through R1 is the only current that is available to turn off an output transistor. By inspecting the circuit, you can see that the voltage across R1 will always be approximately 1.5V, regardless of the amount of current flow through R2 or R3 into the load, as long as neither Q1 nor Q2 goes into cutoff (which is not supposed to happen).

Consider the situation where 5A is being sourced into the load by Q3, and the circuit is trying to turn off Q3. Emitter resistor R2 will have a drop of 1.65V, but this will not cause an increase across R1. Indeed, it merely produces a greater reverse bias voltage across the already turned-off output transistor Q4. When the driver circuit is working properly in its linear region, R1 will always have about 1.5V across it, and the available turn-off current will only be 10mA.

Doug also stated that the approximate conjugate matching of the exponential Vbe characteristics of the upper and lower bipolar output transistors allows the selection of a bias current that nearly eliminates output stage "gain wobbles" through the crossover region as output current changes. This might be true were it not for the very necessary inclusion of the emitter ballast resistors R2 and R3, which effectively ruin this relationship. In other words, you can't just "butt" these two characteristics together and be left with any thermal stability. The combination of the emitter's dynamic characteristic and the series emitter resistor results in a conductance characteristic that increases with increasing current and then asymptotes to the value of the emitter resistance.

The dynamic source characteristic of a MOSFET behaves in almost the same way: its conductance (gm) increases with increasing current and then asymptotes to the value of Rds-on. What counts in a properly-biased emitter-follower or source-follower output stage is the device transconductance versus current, not device transconductance versus base or gate voltage. The main difference between the bipolar and the MOSFET is that the MOSFET's transconductance doesn't approach that of a bipolar until its current is about ten times that of the bipolar. So if you bias the MOSFET output stage at about five to ten times that of a bipolar, you will get a very similar, and equally small, gain wobble through the crossover region. It will, however, be stretched out over a correspondingly larger range of output current swing.

In other words, the quasi Class-A overlap region of the MOSFET output stage is quite a bit larger than that of the bipolar. This is a good thing. It reduces the "suddenness" of the crossover gain wobble and results in softer, lower-order, crossover distortion components.

Of course, as I mentioned in my previous letter, the price to be paid for the MOSFET stage is the significantly higher idle power dissipation. A MOSFET output stage in a 100W amplifier with ±45V rails will dissipate about 45W when biased at 500mA.

It's true that paralleled MOSFETs need to be matched to a degree, but I have never found this difficult. Most of the MOSFETs in a batch tend to be like peas in a pod these days. Matching the devices to within 20 percent of current at the bias current value is more than adequate.

Perhaps the manufacturer Doug was referring to was trying to match the old-style Hitachi MOSFETs. It must have been a killer amp if they needed to put 16 devices in parallel! No paralleling of devices is needed for amplifiers up to about 100W at 8Ω. An IRFP240/IRFP9140 pair, at less than $3 apiece, is plenty adequate for a 100W amplifier given reasonable heatsinking.

Bob Cordell
Holmdel, N.J.

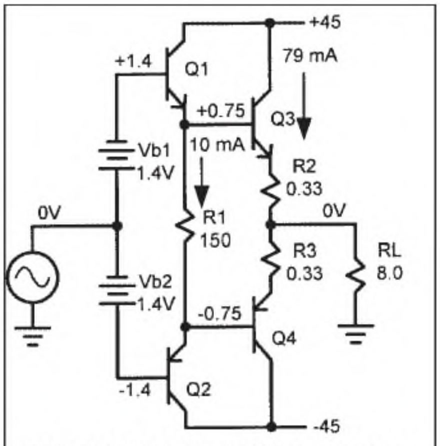

FIGURE 1: Cordell's 100W Type II stage.

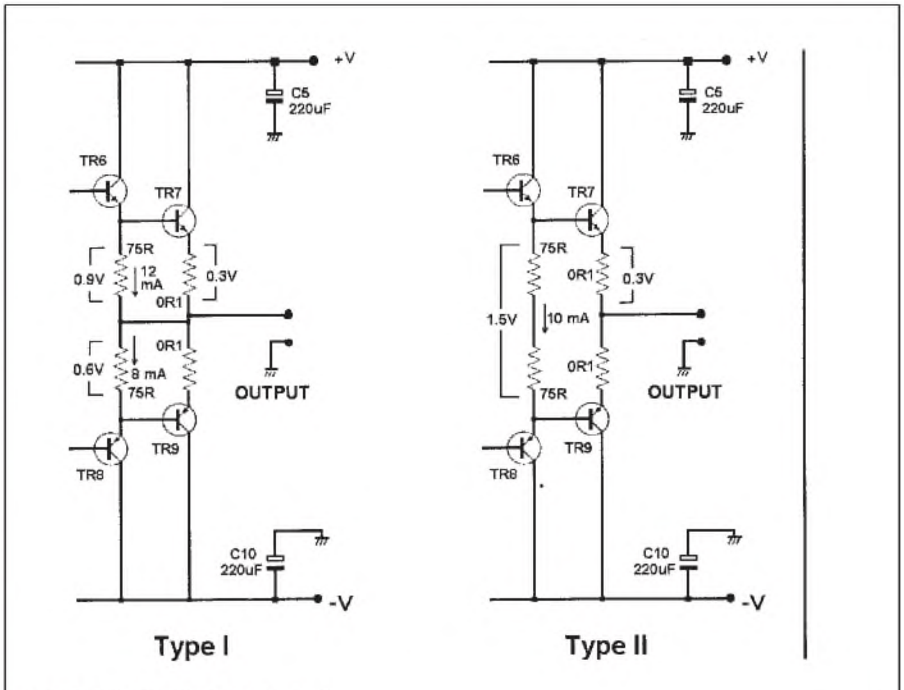

FIGURE 2: Output stage turn-off.

Douglas Self responds:

1. Turn-off in EF output stages. In the example Bob gives, the upper output device is sourcing 5A, so the lower output device is already well turned off, and the current available to turn it off more is not a very useful quantity. Figure 2 shows the situation in the two types of stage when the lower device is on the point of turning off. I have used the same bias conditions of 1.5V between the driver emitters, and all Vbe drops are assumed to be 0.6V. The driver emitter resistor is split into two 75R resistors for clarity.

In the Type II circuit there is always 1.5V across R1, so the current available for turn-off is always 10mA, as Bob says. However, in the Type I, the extra connection to the output rail means that the 1.5V is no longer equally split across the two 75R resistors. The bottom one can have only 0.6V across it at turn-off, reducing the turn-off current available to 8mA. This is the point I was making.

I would not dream of claiming that this 25% difference in turn-off current is the distinguishing feature between a good and a bad amplifier, or anything like that. In most cases the HF distortion will be composed mostly of crossover products. However, the Type II stage does, in my experience, give slightly lower HF distortion in the 10–20kHz region, by an amount that depends on transistor type and is not very designable. It does, of course, also save a resistor!

2. On FET vs bipolar crossover behavior, I think we do differ.

While I agree with most of what Bob writes, I part company when he says, "if you bias the MOSFET stage at about five to ten times that of a bipolar, you will get a very similar, and equally small, gain wobble through the crossover region." This is not my experience. Both in simulation and real life, I have found that FETs turn on much more abruptly, giving sharp gain changes through the crossover region; looking at the THD residual of mainstream FET amplifiers shows a lumpy residual that fits well with this view. The critical difference appears to be that FETs turn on with something like a square-law, whereas bipolars inherently turn on exponentially, and therefore conduction starts very gradually. Whether this applies to all varieties of FETs under all conditions, I would not like to say.

I quite agree that the more spread out the gain wobble the better, giving lower-order distortion products that are better linearized by negative-feedback. Spreading out the wobble can be enhanced with bipolars by using low emitter-resistor values—0R1 seems to give quite adequate quiescent stability—or the simple expedient of putting more transistors in parallel. Both techniques increase the quiescent current, which smoothes out the gain wobble. I go into this in some detail in the third edition of the "Audio Power Amplifier Design Handbook" (available from Old Colony Sound Lab, 1-888-924-9465).

HEATER SUPPLIES

I read with great interest Mark Kelly's article, "Direct Heaters: A Guided Tour." During the past few years I have constructed a couple of single-ended amps using direct heated triodes (DHT). Since the filaments of DHTs carry signal current, the quality of the heater power obviously is of great importance for the sound quality.

I have tried a few different types of regulated DC heater supplies, including some of those presented in the article, and the author confirms my experience. While I may not fully agree with him on the counter-productiveness of current-regulated heater supplies, I really appreciate his article and the experience he shares with us.

Unfortunately, one subject of great interest to me is not mentioned. Because the regulators presented in the article are designed to address differential-mode noise, they are normally ineffective against common-mode noise. Valves may not be very sensitive to differential-mode noise on their heater supplies, but they certainly are sensitive to common-mode noise. Indeed, a DHT filament is as sensitive to common-mode noise as is its grid.

My DHT amps are equipped with series voltage regulated DC heater supplies, but I have added common-mode filtering using series chokes and shunt capacitors. While the sonic results are very pleasing, I am still searching for a regulated heater supply eliminating common-mode noise.

One obvious solution is to try dual voltage regulators connected symmetrically with their ground legs acting "center-tap." This is easy for 2.5V and 10V filament tubes, since 1.25V and 5.0V regulators are readily available. High current regulators for other voltages require more effort and higher parts counts, however. The sonic results are excellent.

One other, more direct, solution is to use separate heater supply transformers, with electrostatic screening between their windings, which would stop the noise right at the source. Such transformers seem not very easy to find, though.

I would appreciate it very much if Mark Kelly would share his experience on the importance of common-mode noise rejection in valve heater supplies.

Lennart Jarlevang
Gothenburg, Sweden

Mark Kelly responds:

Mr. Jarlevang's letter touches on the issue of constant-current sources (CCS), and I have changed my views on this subject since writing the original article, so I will address this first and then talk about the issues raised in the letter. While I still think that voltage rather than current should be the primary control parameter for V/T heaters—and I still like shunt voltage regulators—I now favor feeding the shunt regulator from a constant-current source. The CCS can be either of the two circuits shown in the article. I have also abandoned the use of the LM431 shunt regulator IC in favor of discrete regulation (Fig. 1).

I agree that common-mode noise is an issue with DHT heaters, but I disagree that his scheme with dual regulators is an effective cure. First, assume that out differential mode regulation is close to an AC shunt between the +ve and –ve heater supplies. In the case of fixed bias V/Ts, one side of the heater supply is grounded, so any common-mode noise must by definition appear on both the +ve and earth line. Noise on the earth line is a grounding issue. With the dual regulation scheme the center point of the regulators is still the earth return, so if the earth line is noisy there would appear to be little advantage.

In the case of cathode biased V/Ts, the heater supply is assumed to be floating, and common-mode noise is an issue due to the relatively high impedance to ground of the cathode bypass capacitor. In this case the dual regulators are also ineffectual in that they must be referenced to each other; the ground reference for AC is still the cathode bypass capacitor as before.

In my experience the best solution to common-mode noise problems is battery supply. I am about to implement this for my 211 heaters (requiring 90 odd kg (200 lb) of batteries). I'll keep you posted how it goes.

HELP WANTED

I am trying to find small PCB mount toroid transformers for some projects. I

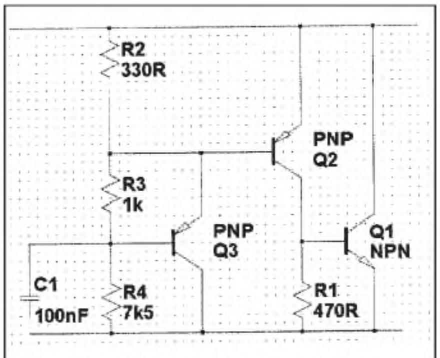

FIGURE 1: Discrete shunt regulator.

noticed in the Sept. 2002 issue ("DASP 2002," p. 4) there was a small toroid. Is there a company that makes those? What are the specs (gauss, pri/sec V, amp), in case I need to make one?

Greg Morgan
gemorga@qwest.com

Edward T. Dell responds:

I do not know of a company that provides small toroidal transformers, but some reader may know of such. We have included your e-mail address and ask such readers to let you know. I suspect that such devices fall into the category of custom production. Many companies supply the cores, although I have not checked the US catalogs. The article you cite from our September issue does give adequate instructions for building this particular device, including the part number for the ferrite core from Ferroxcube. I fail to see how, with the correct cores and following the turns count in the parts list, you could go wrong in winding this device yourself.

I saw a review of what looked to be an interesting unit—so interesting that I'd like to find out where to buy one—in a British magazine *HIFI Plus*. It was called the DYNAVECTOR Super Stereo. It seems to be a processor and amp for driving two speakers placed along the side walls in front of the listener and aimed to bounce their direct sound (sort of like a BOSE 901) off the wall behind the front speakers.

The reviewer said that the unit did indeed improve the sound of regular two-channel music, and, while it didn't really produce a home theater surround sound effect with multichannel recordings, it did give an enjoyable sound field—more so than straight stereo without the Dynavector unit. Are any of your readers familiar with this?

Ray Putnam
putnamr.j@juno.com

I am writing from Italy asking for help with a couple of panel loudspeakers that I used for my home hi-fi system years ago. They are Strathern panel loudspeakers, and now I need to replace the sound-absorbent foam in the back of the panels. It is becoming worse and worse, ruined because of the dry air in my house. I contacted the dealer where I bought the loudspeakers and he said the Strathern factory is closed. Do you know whether this is true, and if so, how I can find a substitute?

Massimo Giuliacci
m.giuliacci@virgilio.it

Let's hope someone knows of a source for replacement foam for your drivers. Foam generally has a relatively short life both as damping panels or driver surrounds. I would also suggest you look for vendors of insulation material and try various alternatives. Rolls of sheet foam are not usually very expensive, and if you can find a similar thickness, it might be an approximate answer. Modify only one of the drivers and compare sound with the new material to the old.–Ed.

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

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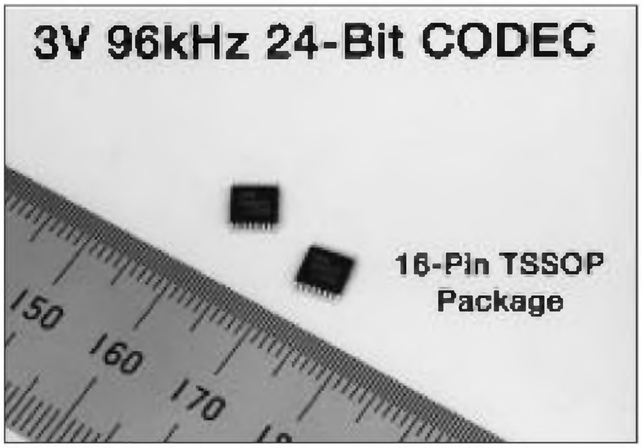
AD INDEX

New Chips on the Block

AKM AK4395 and AK4552 Codec

By Charles Hansen



AKM AK4395

AKM has introduced the AK4395 high-performance digital filter 192kHz 24-bit DAC, aimed at high-end audio systems, including DVD/DVD-audio players, AV amplifiers, and professional audio products. The 28-pin VSOP pin package is pin compatible with the earlier AK4393 and AK4394, to allow easy upgrading for improved performance. The AK4395 adds a 127dB attenuator with 0.5dB resolution, improved 8× digital filter, and lower passband ripple and stopband attenuation.

The analog section provides differential outputs that can drive an AC-coupled 600Ω balanced, or 1kΩ DC-coupled load to +20dBm. A simple balanced low-pass filter is all that is needed to complete a pro-audio interface. The soft mute ramps in 1024 samples between 0dB and full mute for source switching without interruption.

Key AK4395 features:

128× oversampling: 16-, 20-, and 24-bit audio
Sampling rate up to 192kHz
High-performance 24 bit 8× digital filter (slow rolloff option)
Ripple: 0.0002dB, attenuation: 110dB
High tolerance to clock jitter using switched capacitor (SC) technology
Low distortion differential output
Digital de-emphasis for 32, 44.1, and 48kHz sampling
Channel independent digital volume with soft-transition
Soft mute
THD+N: –100dB
DR, S/N: 120dB
I/F format: MSB justified, 16/20/24 bit LSB justified, I^2S
Master Clock:
Normal speed: 256fs, 384fs, 512fs, or 768fs
Double speed: 128fs, 192fs, 256fs, or 384fs
Quad speed: 128fs or 192fs
Power supply: 5V DC
TTL level digital I/F
Small package: 28-pin VSOP pin-compatible with AK4393/4

AK4395 pricing $6.62 US (1k pieces)

AKM AK4552 CODEC

The AKM AK4552 stereo codec gives designers 96kHz, 24-bit resolution in an ultra-small 16-pin TSSOP package. The device is designed to enhance the performance of portable audio products, electronic musical instruments, and car audio systems, with its wide temperature range of –40 to +85°C, and a combined power consumption of only 60mW for the A/D and DAC.

HPF for DC-offset cancel
Single-ended ADC
- S/(N+D): 89dB @ VA = 3.0V
- Dynamic range, S/N: 97dB @ VA = 3.0V

Single-ended DAC
- Digital de-emphasis for 32kHz, 44.1kHz, and 48kHz sampling
- S/(N+D): 90dB @ VA = 3.0V
- Dynamic range, S/N: 100dB @ VA = 3.0V

Audio I/F format: MSB first, twos complement
- ADC: 24-bit MSB justified, DAC: 24-bit LSB justified

Sampling rate:
- 8kHz to 50kHz (normal speed)
- 50kHz to 100kHz (double speed, double speed monitor)
- 100kHz to 200kHz (quad speed monitor)

Master clock:
256fs, 384fs, 512fs @ normal speed
256fs or 384fs @ double speed
128fs or 192fs @ double speed monitor
64fs, 96fs, 128fs, or 192fs @ quad speed monitor
Power supply: 2.4 to 4.0V
Power dissipation: 42mA
Tamb = –40 to 85°C
Very small package: 16-pin TSSOP

AK4552 pricing $2.99 US (10k pieces)

AKM Semiconductor, contact Richard Kulavik toll-free at 888-256-7364, or icinfo@akm.com for info and engineering samples. ❖