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Build This 250W Tube Amp

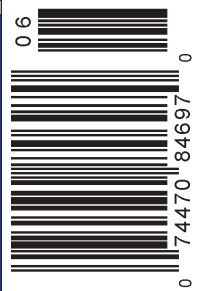
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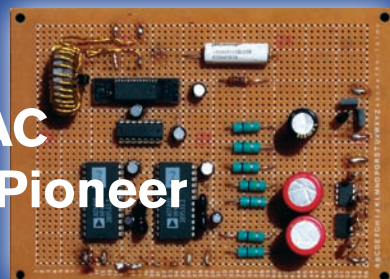


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Mad Katy

A 125W per channel KT88 Amplifier – Part 1

Meet Miss Katy, a stereo tube amp featuring KT88s, who seems to possess the perfect temperament to faithfully reproduce music.

By Dennis Colin

Mad Katy is a 250W (stereo or monoblock) tube amplifier on a 17" x 22" chassis; weight is about 70 lbs (**Photos 1-6**). It has excellent sonic fidelity, transparency, and solidness up to clipping level, achieved by circuit features designed to minimize high-order distortion, reactive load sensitivity, and compressive gain modulations. Ruggedness and reliability are such that it can be used as a musical instrument amp.

To provide 250 sustained watts (1% THD) from a 70 lb. tube amp required

the high efficiency of push-pull Class AB topology (and the power-to-weight efficiency of toroidal transformers). However, the potential anomalies of Class AB operation have been avoided by means of some of the circuit features (**Fig. 1**).



TOPOLOGY AND DESIGN FEATURES

1. Forward (non-feedback) linearizing of output stage transfer function with complementary-transfer driver (Figs. 1 and 2).

2. "Partial Triode" KT88 mode: 67% screen tapping maintains triode-like

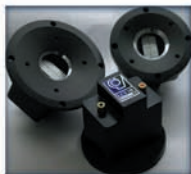
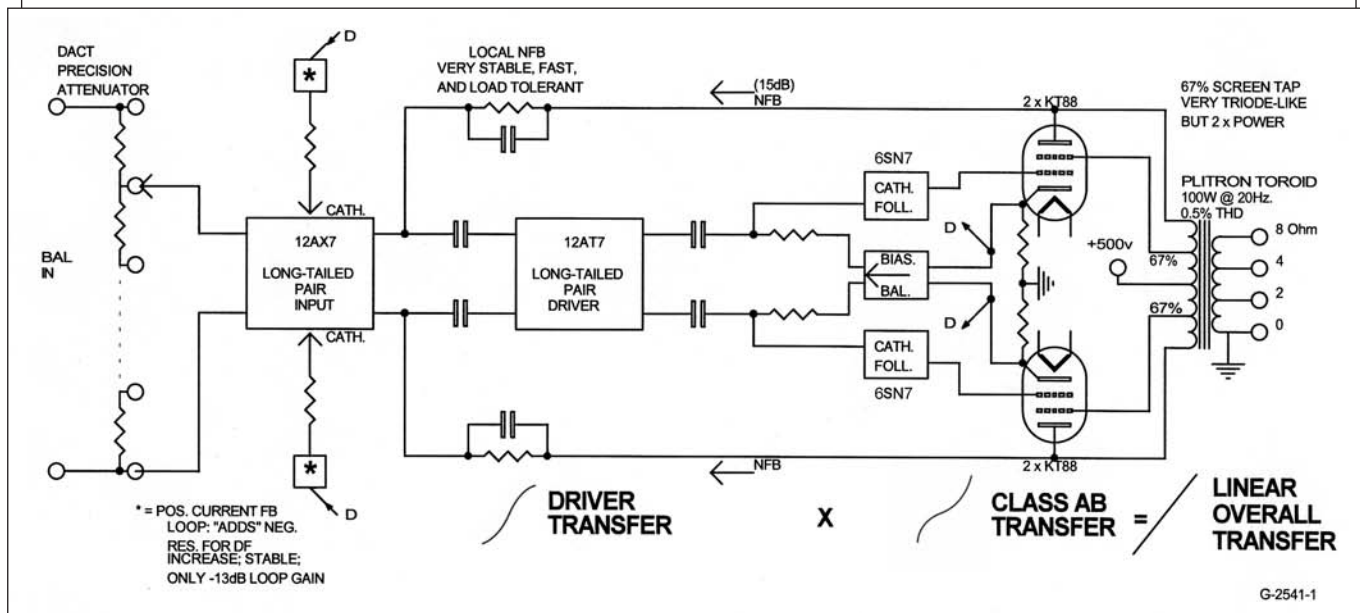
characteristics (Fig. 3), but with twice triode power output.

3. Negative feedback (from KT88 plates to 12AT7 driver input) includes neither the output transformer nor the 12AX7 input stage. With phase shift thereby minimized, stability and tran-

sient response are uncompromised (Fig. 4), and distortion is minimized with (real-world) reactive loads.

4. 6SN7 cathode-follower driver buffers are direct-coupled to the KT88 grids. The low-impedance drive without coupling caps provides wide bandwidth

FIGURE 1: Topology. Forward linearizing: 0.27% THD, 1kHz, 100W, mostly 2nd and 3rd harmonic.



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Frequency Response: 7K-100KHz (-18dB/oct crossover inside)
Crossover Fc: 10KHz/96dB/2.83V/1M (300W)
13KHz/92dB/2.83V/1M (300W)
Impedance: 8 ohm / 12 ohm Ribbon Size: 15 x 25 x 0.006mm



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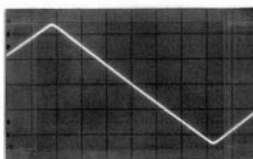
and smooth, anomaly-free overdrive onset and recovery (Fig. 5).

5. Fully-balanced (differential) circuit-

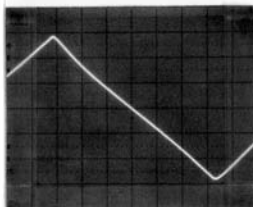


PHOTO 6: Right input.

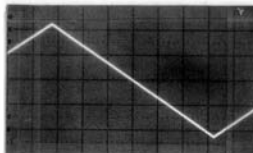
FIGURE 2: Voltage transfer functions.
Source = 1kHz linear triangle wave,
level for $\pm 28.3V$ (200W Pk, 4Ω), 67W
average (100W if sine).



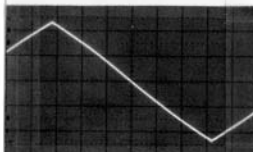
KT88 V_{g-g} driven
from 50Ω source
through Hammond
1650T as step-up.
Each $V_g=0$ to $-120V$.



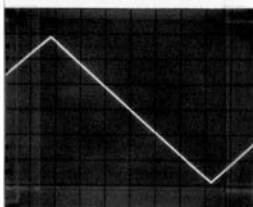
Output voltage
across 4Ω load.
Expanding curvature
is transfer function
of output stage
(4 x KT88, 67% UL,
HV $\approx +500V$, Bias =
50mA/KT88)



Normal
amplifier input
= $\pm 1.095V$ pk
= 0.632V RMS
(0.775V if sine
w/same peaks)



KT88 V_{g-g} (0 to
 $-120V$ each V_g)
compressive
curvature (from
12AT7) compen-
sates output stage



Output voltage
across 4Ω load
(with sine wave,
THD = 0.27% @
100W)

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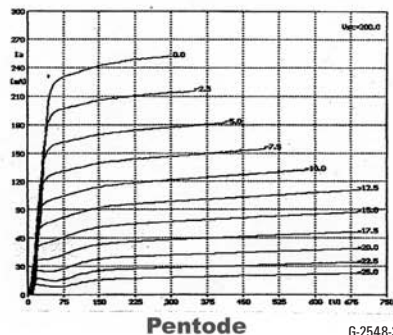
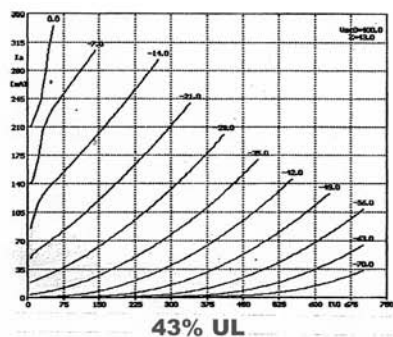
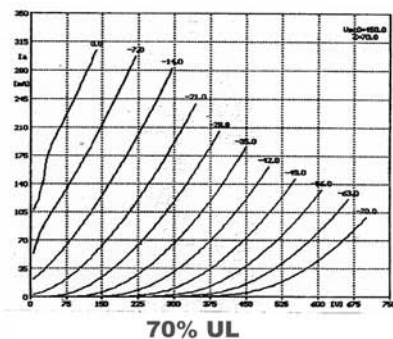
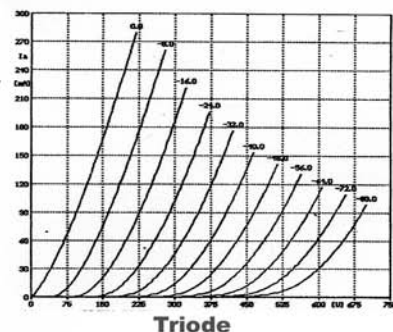
ry throughput provides 113dB "A" weighted dynamic range.

6. Damping factor is adjustable from 4, through essentially infinity, to a small negative output resistance.

PHOTO 7: JJ/Tesla.
KT88.



FIGURE 3: KT88 characteristics.



G-2548-3

This is provided by a local current feedback loop that doesn't compromise stability or transient clarity.

7. Non-feedback bias stabilizer: This, along with extremely solid power supplies, keeps high-power signal compression to less than 0.1dB (**Fig. 6**).

8. Servo DC transformer balance circuitry; no adjustment needed, no saturation even with one output tube inoperative.

9. Plitron toroidal output transformers deliver 112W per channel at 20Hz, 1% THD, and a BW -1dB of 12.5Hz to 39kHz (**Fig. 13**).

10. Ruggedness: output can be shorted during sustained 10dB overdrive, without blowing even a fuse.

ADDITIONAL FEATURES

1. Switchable RCA or balanced XLR inputs.
2. DACT precision step attenuators.
3. Polarity-inverting switches, useful for bridged monoblock.
4. Bias-indicating LED comparators.
5. Six power supply fuses with corresponding status LEDs.
6. Ultra-clean power supplies using five toroidal transformers. All tube heaters and HV are surge-limited. Power-on HV

has 20 second delay. HV FET switch provides "soft" (inaudible) standby/active switching.

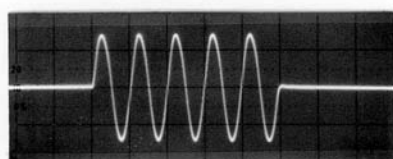
7. AC line is fed through an EMI filter receptacle, inrush current limiter, and voltage transient suppressor.

Overall, the amplifier (schematics, **Figs. 7, 12, 17 (in part 2)**) appears complex, with a fairly large parts count. However, the audio path is straightforward and relatively simple, as it should be. Much of the complexity is due to the "housekeeping functions" and six very solid power supply voltages (four regulated).

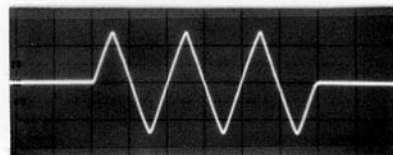
Materials cost was about \$2300. In the interest of economy, you could use the basic amp with simpler supplies and without the "housekeeping" functions. You could use 6550 output tubes, but

FIGURE 4: Tone bursts. 200W peak, average power stated. 4Ω load.

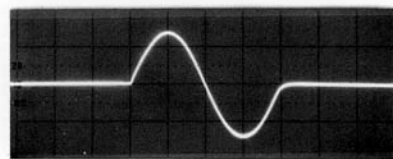
20kHz 100W sine • 10kHz 68W triangle
3kHz 1-cycle 100W sine • 80Hz 100W sine
40Hz 100W sine (envelope delay due to 6Hz -3dB rolloff)



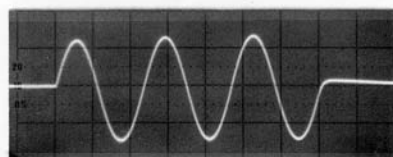
20kHz 100W sine



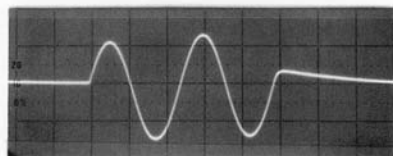
10kHz 68W triangle



3kHz 1-cycle 100W sine



80Hz 100W sine



40Hz 100W sine (envelope delay due to 6Hz -3dB rolloff)

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I strongly recommend JJ/Tesla KT88s (Photo 7). They provided 10% more power and lower distortion than KT88s from another high-quality manufacturer.

Table 1 summarizes the measured performance data.

AMPLIFIER CIRCUIT

Refer to the amp circuit in **Fig. 7**. RCA or balanced XLR inputs are selected through S1; C1 removes any DC offsets that would cause attenuator transients. The signal is then applied to AT1, a DACT precision step attenuator. R6 is to (a) taper this for about 1dB steps near maximum gain (attenuator itself is 2dB/step down to -34dB), and (b) lower the maximum output R seen by the 12AX7 input tube V1; this is to reduce variation of Miller effect loading. R7 is to balance the average (over attenuation range) source R to the two polarity input triodes, to maximize CMRR.

S2 is a polarity inverting switch, facilitating use as a 250W bridged monoblock.

V1 is a long-tailed pair with excellent output balance and input CMRR, because the cathodes are supplied from a high (68k) resistance, R14, and the -200V bias supply; compared to the maximum $\pm 1.2V$ input swing, this is a nearly ideal (and linear resistive) current source. Zeners D1, 2 provide a path for this current in standby (where the -200V stays on) to prevent the grids from conducting significantly. (The -200V stays on to prevent output tube bias surges and thumps when switching out of standby.)

Note that the negative feedback (NFB1, NFB 2) from the KT88 plates is fed (through R18, 19, C6, 7; R28, 29, C10, 11) to the 12AX7 plates. This lowers the load resistance seen by the 12AX7 to about 7k Ω each plate, a very low value. However, only about 9V p-p is

needed for full output; distortion is very low. Furthermore, what little there is, is symmetrical and compressive, adding slightly to the intentional compression of the 12AT7 driver; this is used to linearize the KT88 Class AB output, which has a complementary expansion.

C8/R25 and C9/R26 form the dominant LF rolloff pole, at around 28Hz open loop. This ensures LF stability, while the 15dB of NFB results in an overall LF -3dB rolloff of 6Hz.

R16 and C4 provide a small amount of HF flatness compensation. P1 is used for fine gain match of the two channels.

DRIVER

This is a 12AT7 long-tailed pair (V2). D3 provides the same function as D1, 2; a zener is not needed here because the grids are biased at about +5V from the symmetry trimpot P2, so the cathodes are at about +9.5V DC, keeping D3 off except in standby.

The 12AT7 driver has a smoothly compressive limiting curve that very well compensates the expanding transfer function of the output stage (**Fig. 2**).

V3 (6SN7 GTB) is a dual cathode follower, DC coupled to the KT88 grids. This provides the advantages of

TABLE 1: MEASURED PERFORMANCE DATA

Amplifier data 4 Ω load 117V 60Hz AC line						
Maximum power (3% THD)						
1kHz: 142W (1 channel), 268W (combined or monoblock), 155W per channel						
tone burst						
20Hz: 112W per channel 20kHz: 90W per channel						
THD %						
	1W	10W	25W	50W	100W	IMD, 200Hz + 10kHz, 4:1
20Hz	0.05	0.14	0.22	0.33	0.50	0.7%, 100W
1kHz	0.03	0.08	0.12	0.18	0.27	0.2%, 70W
5kHz	0.05	0.16	0.22	0.51	0.90	1.0%, 70W, 4 Ω //2 μ F load
10kHz	0.09	0.32	0.55	1.1	1.7	0.09%, 30W
20kHz	0.18	0.65	1.4	2.4	3.3	

Frequency response: -1dB BW = 12.5Hz - 39kHz; -3dB BW = 6Hz - 65kHz.

Rise time (10-90%): 4.0 μ s at 10W, 5.3 μ s at 185W (square-wave power).

Damping factor: >20, 40Hz - 10kHz; 13 at 20Hz; 9 at 20kHz.

Sensitivity: 0.775V RMS for 100W, 1.36V peak for clipping.

Input impedance: 76k Ω (RCA), 16.4k Ω (XLR); C_{IN} = 115pF (all at max gain).

Noise output: 0.053mV RMS, 20Hz - 20kHz unweighted.

Hum output: 0.31mV RMS

"A" weighted signal/noise + hum: 113dB re 125W

Channel separation: 86dB (20Hz - 10kHz), 75dB (20kHz)

Balanced input common-mode rejection: 33dB, 20Hz - 20kHz

AC power consumption:

154W, standby 398W, active, no signal

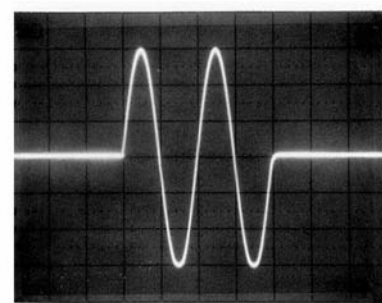
677W, 2 $\times$ 100W sine

915W, 2 $\times$ 200W square

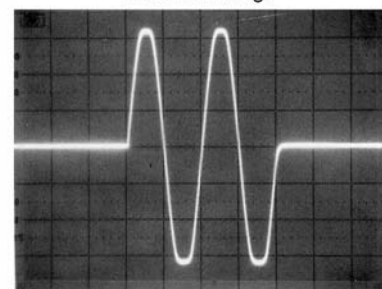
500W, typ. music at full power

AC line input range: 110-125V RMS, 60Hz

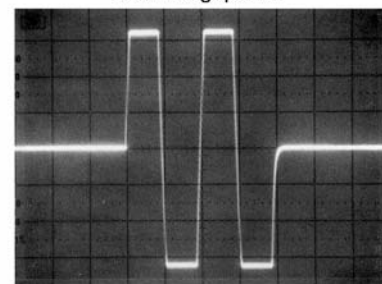
FIGURE 5: Bridged monoblock 1kHz 2-cycle burst. 8 Ω load 10% duty 20V/div. 0.5V/div. AC line = 116V RMS 60Hz $\approx$ 1% THD/234W sine average • 1.5dB overdrive/300W average power • 10dB overdrive/452W average power 528W peak power



$\approx$ 1% THD
234W sine avg.



1.5 dB Overdrive
300W avg. power

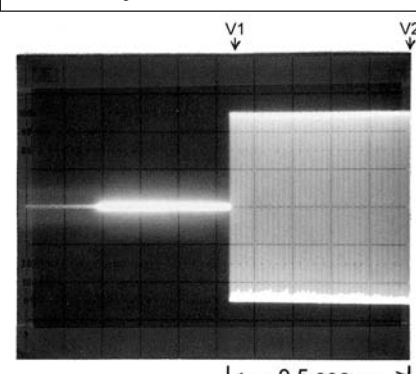


10dB Overdrive
452W avg. power, 528W peak power

G-2548-4

FIGURE 6: Amplitude stability of full-power signal.

1kHz step onset 100W sine 4 Ω load v_2/v_1 = 1.0113. change, 0 to 0.5sec = 1.13% = 0.098dB Amplitude change, 0.5 - 60sec (not shown) = -0.0174dB, both channels, driven, 100W each Amplitude stability, onset of 2 $\times$ 100W signal to 60sec later = $\pm$ 0.058dB.



G-2548-6

strange-looking things near the transformer primary. But first, the cathodes have 10 Ω 1% current-sensing resistors, serving several functions: (a) bias sensing

INPUTS

RCA J1

R1 1M

R2 10K

R3 620

R4 10K

S1 DPDT

R5 39K 1% 1W

R6 39K 1% 1W

R7 6K

R8 47K

R9 47K

R10 47K

R11 1K

R12 5K 2% 1W

R13 5K 2% 1W

R14 68K 2W

R15 1K

R16 2K

R17 470

R18 1K

R19 1K

R20 1K

R21 100K 1% 1W

R22 100K 1% 1W

R23 18K

R24 18K

R25 220K

R26 220K

R27 27K 2W

R28 360K 2W

R29 360K 2W

R30 1K

R31 33K 2W

R32 33K 2W

R33 100K

R34 330K

R35 18K 1% 1W

R36 18K 1% 1W

R37 1K 1/2W

R38 15K 5W

R39 1K

R40 1K

R41 1K

R42 1K

R43 10 1% 1W

R44 10 1% 1W

R45 10 1% 1W

R46 10 1% 1W

R47 100

R48 100

R49 100

R50 100

R51 1K 1% 1W

R52 1K 1% 1W

R53 1K 1% 1W

R54 1K 1% 1W

R55 200 5W

R56 200 5W

R57 200 5W

R58 200 5W

R59 300 5W

R60 5W

R61 1K 5W

R62 1K 5W

R63 1K 5W

R64 1K 5W

R65 1K 5W

R66 1K 5W

R67 1K 5W

R68 1K 5W

R69 1K 5W

R70 1K 5W

R71 1K 5W

R72 1K 5W

R73 1K 5W

R74 1K 5W

R75 1K 5W

R76 1K 5W

R77 1K 5W

R78 1K 5W

R79 1K 5W

R80 1K 5W

R81 1K 5W

R82 1K 5W

R83 1K 5W

R84 1K 5W

R85 1K 5W

R86 1K 5W

R87 1K 5W

R88 1K 5W

R89 1K 5W

R90 1K 5W

R91 1K 5W

R92 1K 5W

R93 1K 5W

R94 1K 5W

R95 1K 5W

R96 1K 5W

R97 1K 5W

R98 1K 5W

R99 1K 5W

R100 1K 5W

C1 2.2uF 50V

C2 68pF

C3 300V

C4 2N2

C5 2.2uF 630V

C6 5pF 500V

C7 5pF 500V

C8 50N 1KV

C9 50N 1KV

C10 5pF 500V

C11 5pF 500V

C12 2.2uF 630V

C13 470N 630V

C14 470N 630V

C15 47pF 300V

C16 2.2uF 630V

C17 100uF 10V

C18 100uF 10V

C19 100uF 10V

C20 6.8uF 630V

C21 100uF 10V

C22 100uF 10V

C23 100uF 10V

C24 100uF 10V

C25 100uF 10V

C26 100uF 10V

C27 100uF 10V

C28 100uF 10V

C29 100uF 10V

C30 100uF 10V

C31 100uF 10V

C32 100uF 10V

C33 100uF 10V

C34 100uF 10V

C35 100uF 10V

C36 100uF 10V

C37 100uF 10V

C38 100uF 10V

C39 100uF 10V

C40 100uF 10V

C41 100uF 10V

C42 100uF 10V

C43 100uF 10V

C44 100uF 10V

C45 100uF 10V

C46 100uF 10V

C47 100uF 10V

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C111 100uF 10V

C112 100uF 10V

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C132 100uF 10V

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C134 100uF 10V

C135 100uF 10V

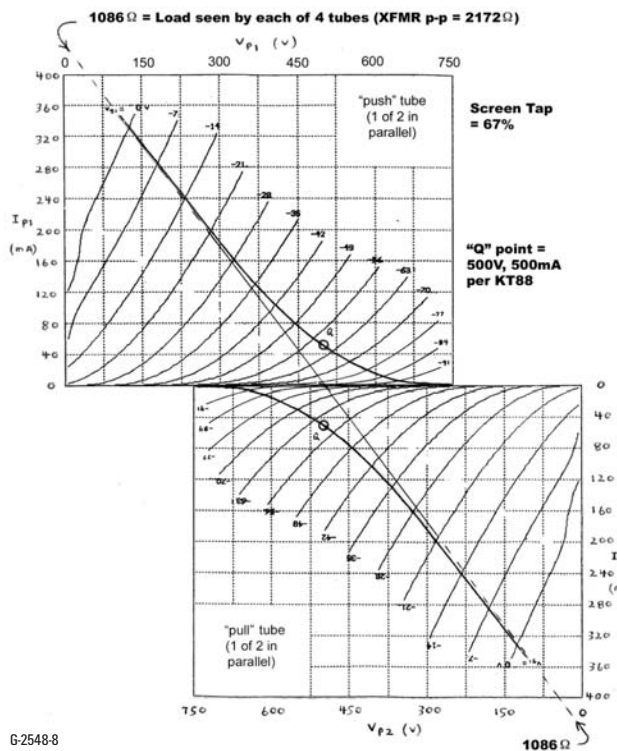
C136 100uF 10V

C137 100uF 10V

C138 100uF

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FIGURE 8: Push-pull Class AB load lines.



G-2548-8

FIGURE 9: PAT-4142-00 transformer primary rewiring. Good match for four KT88s, HV = +500V, bias = 50mA/tube, AB1. Primary windings must be in below order for best leakage inductance symmetry (if using 66 2/3% screens).

Primary winding	Relative turns ratio	DCR
1	1.0000	34.5Ω
2	0.5000	15.8Ω
3	0.5000	15.4Ω
4	1.0000	29.9Ω

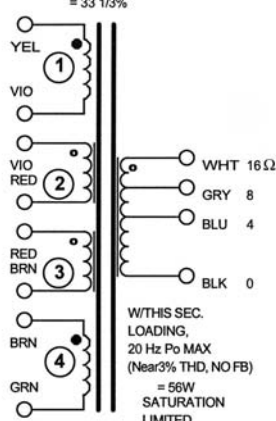
Measured on s/n 200202

Turns ratio, full primary to: Blk-blk = 31.38 Blk-grk = 22.24 Blk-wht = 15.81

Approximate interwinding C: any primary pair 100pF Total primary to sec 440pF.

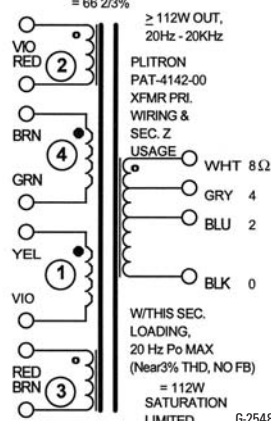
PLITRON ORIG.

SCREEN TAPS = 33 1/3%



AS USED

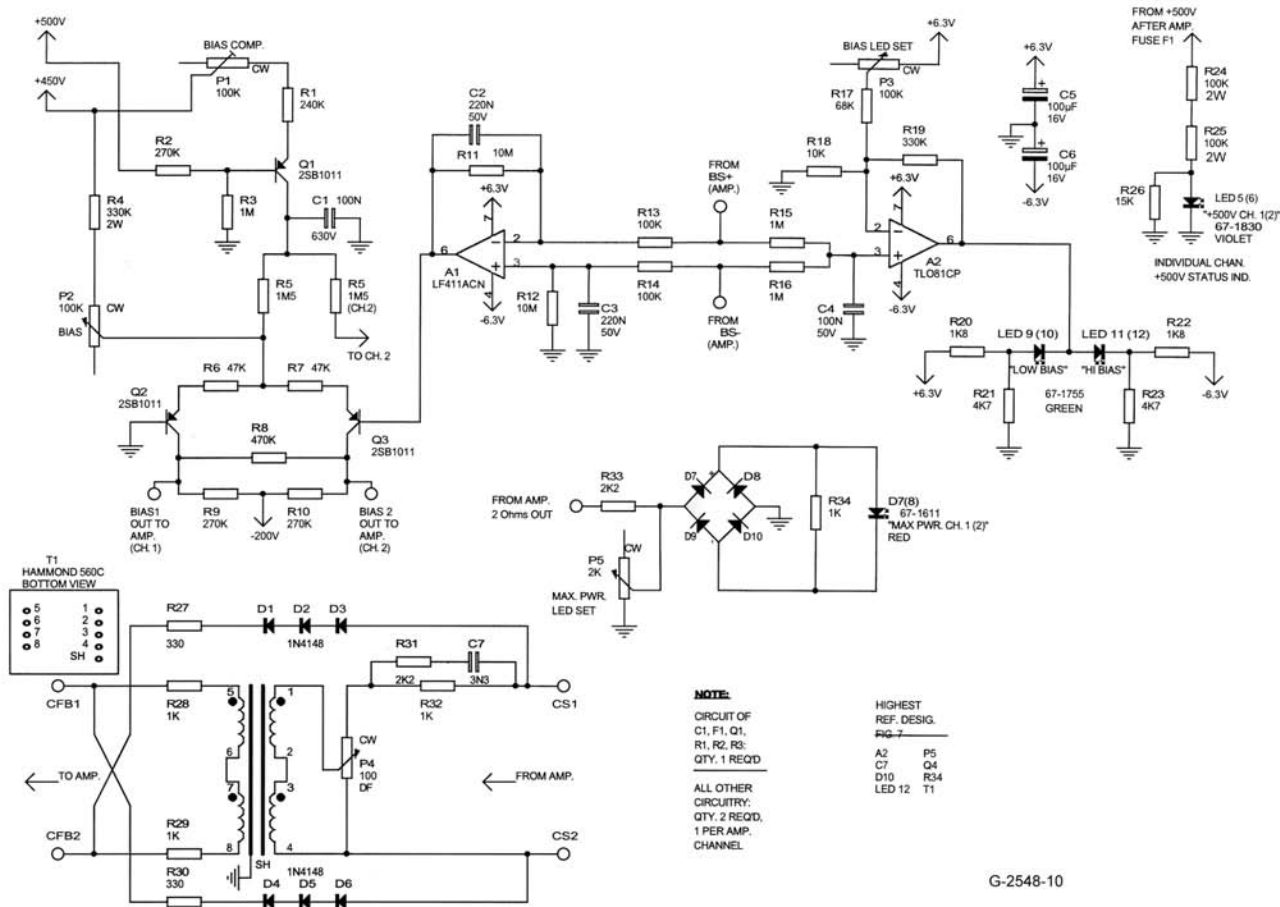
SCREEN TAPS = 66 2/3%



G-2548-9

FIGURE 10: Bias, servo balance, current FB, and status LED schematic.

Note: initial pot settings: P1 = 48k (w-cw) • P2 = Full ccw • P3 = 44k (w - cw) Damping factor current feedback loop and output current limiter. Note: Set P4 (w - ccw) to 40Ω initially. Maximum power ind. Note: with P5 ≈ 575Ω (w - ccw), LED just on at 75W sine out, bright at 100W.



G-2548-10

(at the “BS+” and “BS-” points); these drive the servo-balance circuit, indicating LEDs, and chassis-top banana jacks; then (b) points labeled “CS1” and “CS2” drive a current feedback (CFB) circuit (on **Fig. 10** with the servo balance), returned to the 12AX7 cathodes (points “CFB1” and CFB2”); this is the damping factor compensator.

Finally, the “CS1,” “CS2” points also feed the normally-off current-limiting feedback diodes shown in **Fig. 10** in the event of output shorting or overloading.

Figure 8 shows the composite push-pull load line sensed by each one of the four output tubes.

OUTPUT TRANSFORMER

C18, L2, R59 and C19, L3, R60 were necessary to damp out a strong transformer impedance peak around 450kHz. This had raised mayhem with the NFB loop, until the above components resulted in a wideband smooth capacitive impedance curve. Note that this could not have been avoided by changing to the usual NFB feed from the secondary, because this transformer has significant spurious transmission to beyond

5MHz—Plitron states a wide bandwidth, but it's a little *too* wideband! No problem, fixed it.

The second unusual step was to rewire the primary (**Fig. 9**) for 67% UL screen taps, to have near triode-like operation (more linear, lower output Z) without the severe power penalty of pure triode operation. As shipped, the primary has three leads that are each a pair with ends soldered together; unsoldering these revealed four isolated windings, which I re-configured as shown in **Fig. 9**. Note: don't cut these leads to shorten them; each lead is a number of fine enameled magnet wires—difficult to strip.

Following is an explanation of L1 (with its damping resistor R5): above about 50W at 20kHz, about 5% asymmetry was observed. Reversing push/pull tubes made no difference. Then I measured leakage inductance of each primary half to sec. (by shorting sec.)—one primary half showed 1.0mH, the other 2.2mH. So I added the difference, a 1.2mH 1A inductor (L1) in series with the lower -L primary plate lead. This greatly improved 20kHz symmetry.

TRANSFORMER LOW FREQUENCY CAPABILITY

The transformer is designed for 80W and 4kΩ p-p to 4, 8, 16Ω. The half power LF frequency is stated as 14Hz (40W). Now when saturation-limited, the maximum output follows a 6dB/octave slope (the flux limit allows a given voltage per turn per Hz, so maximum power drops to ¼ for an octave lowering of frequency); then at 20Hz the maximum power would be about 80W (measured 56W maximum in-circuit). I reasoned that, staying within the flux-voltage limit, I could halve the load impedance (hence the 2-4-8Ω outputs), drawing twice the load current, and achieve nearly double the LF power limit. (“Nearly” because now the winding resistances would have double the losses.) Also, now the primary Z pp would match the 2kΩ load of 4-KT88s in 67% UL.

Norman Woo of Plitron thought my ½Z idea should be OK; he's right: the amp can deliver about 115W at 20Hz, and at 100W the THD is only 0.5%. I tried, for curiosity, a Hammond 1650T, rated at 120W with -1dB (95W) at



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30Hz. Not only was 20Hz power much lower than with the Plitron toroid, but distortion was much higher even at power levels well below the maximum.

BIAS COMPENSATOR

Bias magnitude is compensated against supply variations, but is not servo (feedback) regulated; this would be complicated in a Class AB stage because the average cathode current increases nonlinearly with signal power. On the other hand, DC bias *imbalance* is a differential signal like the audio itself; think of it as a “zero-frequency audio” error signal. Therefore, you can apply a linear negative FB (servo) loop to reduce DC imbalance to an arbitrarily low level.

Joseph Norwood Still points out (2/00 *GA*, p. 62) that if both plate and negative bias supplies are unregulated, then as AC line voltage changes, these supplies will track together, tending to maintain reasonably constant tube bias current. But here the negative bias supply is regulated while the plate supply is not. In addition, I wanted the bias current stabilized against plate supply “sag” due to high power loading, to eliminate signal compression.

Referring to **Fig. 10**, Q1 compares the unregulated 500V plate supply with the regulated 450V supply (as a reference), and its collector current increases as the 500V decreases, whether from load sagging or AC line drop. P1 adjusts the amount of this compensation current.

This Q1 current, divided to the two amp channels by the R5 1.5m resistors, is added to the current from the bias-setting pot P2; the summed current feeds one emitter of the diff-amp Q2-Q3. C1, in conjunction with the two channels’ 1.5m resistors, filters the compensation signal above 2Hz.

The diff-amp outputs “BIAS 1” and “BIAS 2” drive the amplifier’s 6SN7 grids through R33 and R34 (**Fig. 7**). Back to **Fig. 7**, if the output of A1 is zero, the two bias outputs are (ideally) equal. An error voltage from A1 serves to differentially unbalance the two bias outputs as needed.

SERVO BALANCE

Straightforward in principle, the design of the negative FB loop is not trivial; the applicable parameters of loop-gain roll-

off and phase margin must be adequate to ensure stability. This must be maintained (and was tested) from zero signal to overdrive, but I also made sure that no instability would occur even with output transformer saturation (which normally won’t occur, but I induced with an overdriven 10Hz signal). Furthermore, the servo-balance loop, which attenuates DC and infrasonic frequencies, introduces a LF rolloff (highpass pole/zero response)

into the overall amplifier NFB loop.

All of this made the design of the simple-looking circuit rather “interesting.” But it works very well, and has proven to be completely stable with a variety of abusive test signals, including highly overdriven bass-heavy music.

The cathode-current sensing voltages “BS+” and “BS-” from the amp are fed to A1, a diff-amp with DC gain of 100. C2 and C3 provide a LF rolloff of 0.07Hz

FIGURE 11: Positive current FB loop generating negative output resistance. A voltage (V2) proportional to load current (i_o) increases amplifier drive (v_a) when load impedance (Z_L) decreases, compensating voltage drop across amplifier output resistance (r_a).

$$V_a = v_1 + i_o r_{sc} \quad V_o = v_a - i_o r_a \quad V_O = v_1 + i_o (r_{sc} - r_a) \quad R_o = \text{output resistance seen by load} \quad R_o = -dv_O/di_o = r_a - r_{sc}$$

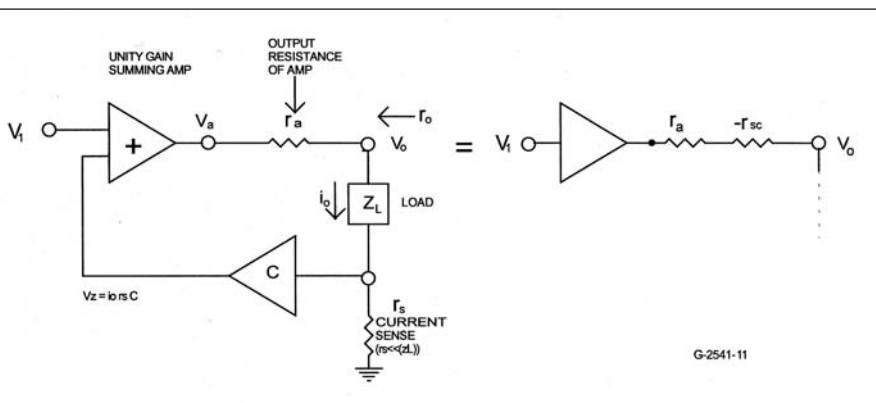


FIGURE 12: Panel component locations. Top, front panel; bottom, rear panel

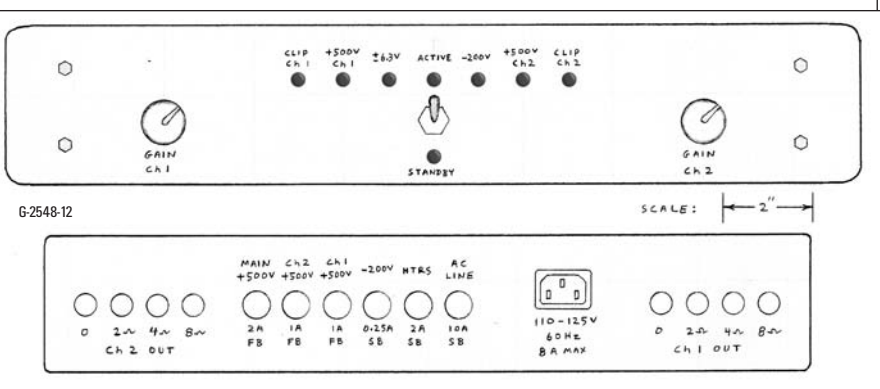
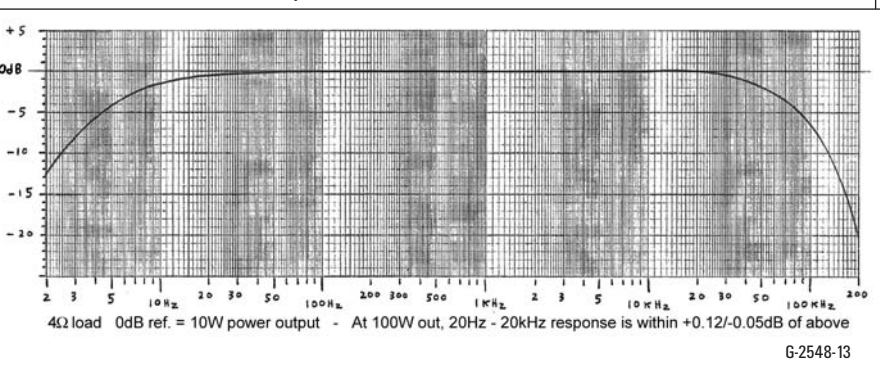


FIGURE 13: Amplifier frequency response. 4Ω load 0dB ref. = 10W power out. At 100W out, 20Hz–20kHz response is within +0.12/-0.05dB of above.



(2.2 sec. time constant); this is the dominant FB loop pole. This also attenuates 20Hz audio signals by 49dB with reference to the DC loop gain. The output of A1 drives the bias diff-amp Q2-Q3, completing the loop.

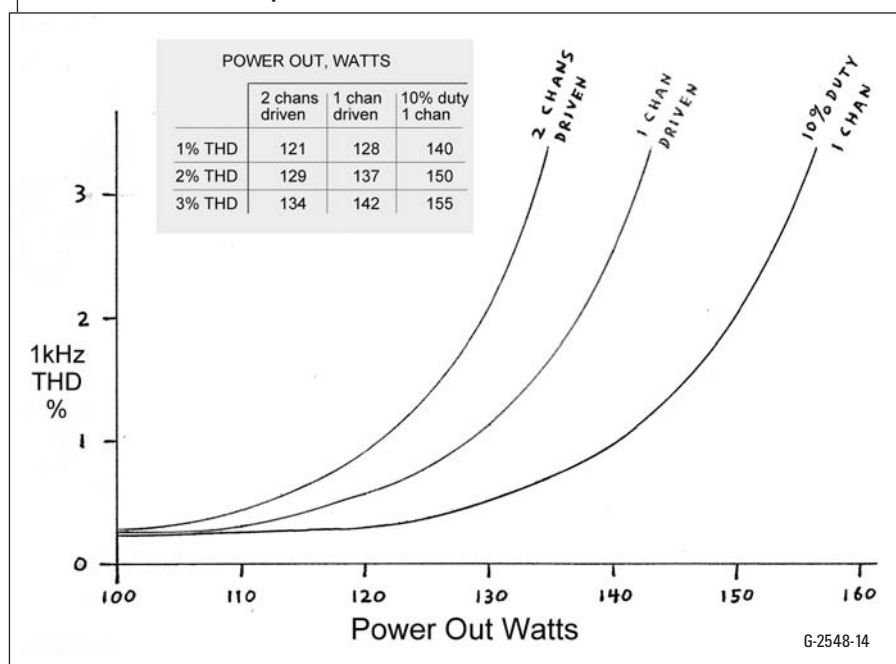
With matched KT88s, quiescent DC imbalance is held to 0.2mA in the transformer primary. With a 100W 20Hz signal, DC imbalance is less than 1mA. With intentionally mismatched tubes, the corresponding values are 2mA and 3mA.

For an extreme (and unintended) test, when I was (at first) using 33% UL tapping, one KT88 screen resistor blew open during an extended overdriven square-wave test (50mA average, 140mA peak screen current—that KT88 permanently became a beautiful blue “neon” glow lamp after replacing the screen resistor (and that tube non-conducting), the servo-balance still held the transformer DC imbalance to 4mA. (Note that with the chosen 67% UL tapping, no amount of overdrive abuse exceeds any KT88 ratings.)

Kevin O'Connor, in his excellent book *Principles of Power*, points out that toroidal push-pull output transformers need much lower DC imbalance current to saturate than conventional E-I

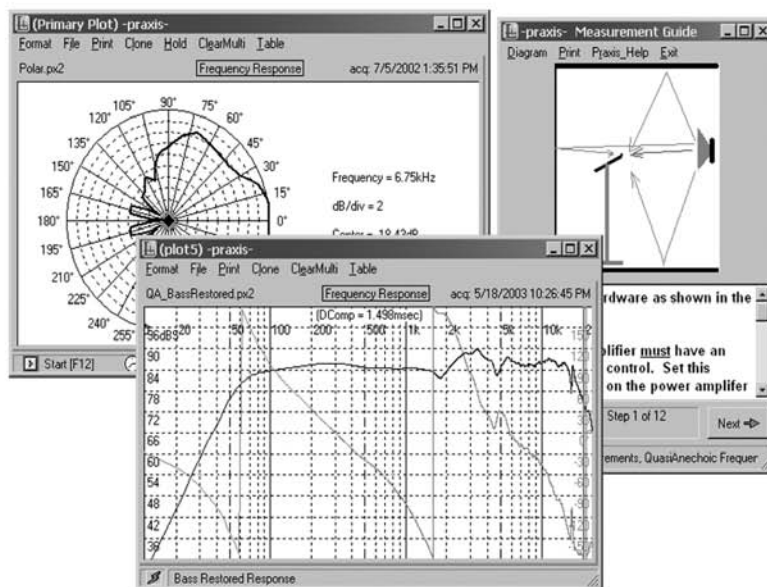
transformers. True—I measured 8mA DC imbalance needed to approximately half-saturate the Plitron PAT-4142-00, while the 120W Hammond 1650T needed 50mA. With this servo-balance

FIGURE 14: THD versus power, 1kHz. 4Ω 117V AC line.



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circuit, no visible (nor audible) saturation occurs even at 100W, 20Hz with moderately mismatched tubes.

BIAS STATUS LEDs

The “BS+” and “BS–” voltages are averaged by R15-16, LP filtered above 3Hz by C4, and amplified by A2 with variable DC offset from P3. A2 output drives a “LED-comparator” network. First, bias is set to 50mA/KT88 (by measuring 0.50V at “BS+” and “BS–”); then P3 is adjusted so the two LEDs glow equally. From then on, the voltmeter is not needed: a 5mA (per KT88) bias change extinguishes the appropriate (“low bias” or “hi bias”) LED, and bias can be adjusted (with P3) to within 1mA by balancing the LEDs’ brightness.

The maximum power indicator is very simple. The circuit with P5 set as shown (575Ω, w-ccw) just starts to turn on the red LED at about 100W sine wave power. At clipping onset (about 135W sine) this high-efficiency LED will flash brightly enough to use as a reading lamp.

DAMPING FACTOR CURRENT FEEDBACK LOOP

I’ve saved the most interesting part of the supplemental functions for last. This uses positive load current feedback (CFB) to cancel part (or all if desired) of the amplifier’s output resistance. Don’t worry; while this is positive FB, the loop gain is only about –13dB, completely stable. A proven old idea, it’s been used in vintage Altec studio amps and (1997) in the Sonic Frontiers Power 2 amp, a 110W per channel stereo tube unit (reviewed in *Stereophile*, May 1997, p. 153).

Transformer primary reflected load current is sensed from the KT88 cathode resistors, summed at the “CS1” and “CS2” points. After some HF comp. (R31-C7) and level control (P4), the differential signal proportional to speaker load current is sensed (with very high common-mode rejection) by T1, a Hammond 560C unit of broadcast quality. As used here, its –1dB bandwidth is about 10Hz–100kHz, with probably less than 0.01% THD. Its output feeds (through R28–29) the 12AX7 cathodes (points “CFB1” and “CFB2”), completing the loop. (Note: Diodes D1 through D6 are not part of the CFB circuit; they

PARTS LIST

Ref.	Description			Mfr.P/N	Dist.	Dist. P/N
AT1	Step attenuator			DACT CT2 100k-1		
C6,7,10,11	Cap, 5pF, 500V	ceramic disc				
C2	cap, 68pF, 100V	ceramic disc				
C18,19	cap, 1.5nF, 1600V					
C4	cap, 2.4nF, 100V	ceramic disc				
C8, 9	cap, 0.022μF, 1kV		Hovland		AES	C-HD022-1000
C13,14	cap, 0.47μF, 630V	poly	Solen		AES	C-FSD47-630
C1	cap, 2.2μF, 250V	poly	Solen		AES	C-FS2D2-250
C5,12,16	cap, 2.2μF, 630V	poly	Solen		AES	C-FS2D2-630
C17	cap, 6.8μF, 630V	poly	Solen		AES	C-FS6DB-630
C3,15	cap, 47μF, 350V	electrolytic	Panasonic	EEU-EB2V470	DK	P5934
1, 2	diode, Zener 10V			1N5240	DK	1N52408DICT
D3	diode			1N4148		
F1	fuse, 1A FB					
J1	jack, RCA					
J2	Jack, XLR		Neutrik	NC3FP-1-B	Newark	B9F152
L1	inductor, 1.2mH, 1A		Hammond	1540M14		
L2, 3	inductor, 100μH, 100mA		Hammond	1530B104		
P1	trimpot, 2k					
P2	trimpot, 10k					
R43-6	Res, 10Ω, 1%	1W	MF			
R47-50	Res, 100Ω, 1%	¼W	MF			
R55-8	Res, 200Ω, 5%	5W	wire-wound		DK	ALSR5J-200
R59, 60	Res, 300Ω, 5%	5W	wire-wound		DK	ALSR5J-300
R17	Res, 470Ω, 5%	¼W	MF		DK	
R3	Res, 620Ω, 5%	¼W	MF		DK	
R5	Res, 1k, 5%	5W	wire-wound		DK	ALSR5J-1.0k
R8,11,30,15,37,51-54,39-42	Res, 1k, 1%	1W	MF			
R16	Res, 2.2k, 5%	¼W	MF			
R20	Res, 4.7k, 5%	1W	MF			
R7	Res, 6.8k	¼W	MF			
R12, 13	Res, 5.6k, 1%	¼W	MF			
R2, 4	Res, 10k, 1%	¼W	MF			
R38	Res, 15k, 5%	5W				
R23, 24	Res, 18k, 5%	¼W	MF			
R35, 36	Res, 18k, 5%	2W				
R27	Res, 27k, 5%	2W				
R31, 32	Res, 33k, 5%	2W				
R6	Res, 39.2k, 1%	¼W				
R14	Res, 68k, 5%	2W				
R21,22,33,34	Res, 100k, 1%	1W				
R25,26	Res, 220k, 5%	¼W				
R18,19,28,29	Res, 360k, 1%	2W				
R9, 10	Res, 470k, 5%	¼W				
S1, S2	switch, DPDT (2)					
T1	output transformer				Plitron	PAT-4142-00
V1	Vacuum tube			12AX7	AES	
V2	vacuum tube	12AT7	AES			
V3	vacuum tube			6SN7GTB	AES	
V4-7	vacuum tubes			(JJ) KT88	AES	
A1	op amp			LF411ACN		
A2	op amp			TL081CP		
C7	cap, 3300pF, 100V					
C4	cap, 0.1μF, 50V					
C1	cap, 0.1μF, 630V	▽				
C2, 3	cap, 0.22μF, 50V					
C5, 6	cap, 100μF, 16V	electrolytic				
D1-10	diode			1N4148		
LED5 (6)	LED, violet				Digi-Key	67-1830
LED7 (8)	LED, red				Digi-Key	67-1611
LED9(10) 11(12)	LED, green				Digi-Key	67-1755
P1	pot, 100k Lin	2W Precision		RV4LAYS A 104A	Digi-Key	RV4L104C
P2, 3	trimpot, 100k				Digi-Key	3296Y-104
P4	trimpot, 100Ω					
P5	trimpot, 2k					
Q1-3	transistor, PNP	▽ (Q1)		2SB1011		
R27, 30	Res, 330Ω, 5%	¼W				
R28,29,32,34	Res, 1k, 5%	¼W				
R20, 22	Res, 1.8k, 5%	¼W				
R31, 33	Res, 2.2k, 5%	¼W				
R21, 23	Res, 4.7k, 5%	¼W				
R18	Res, 10k, 5%	¼W				

Continues→

provide current limiting negative FB and are normally out of the picture.)

So for a given amplifier input voltage, lowering the load impedance draws more current, which feeds back more (in-phase) drive voltage to the 12AX7 cathodes. This cancels an adjustable amount of the normal output voltage drop with lower load impedance, effectively "adding" a negative resistance.

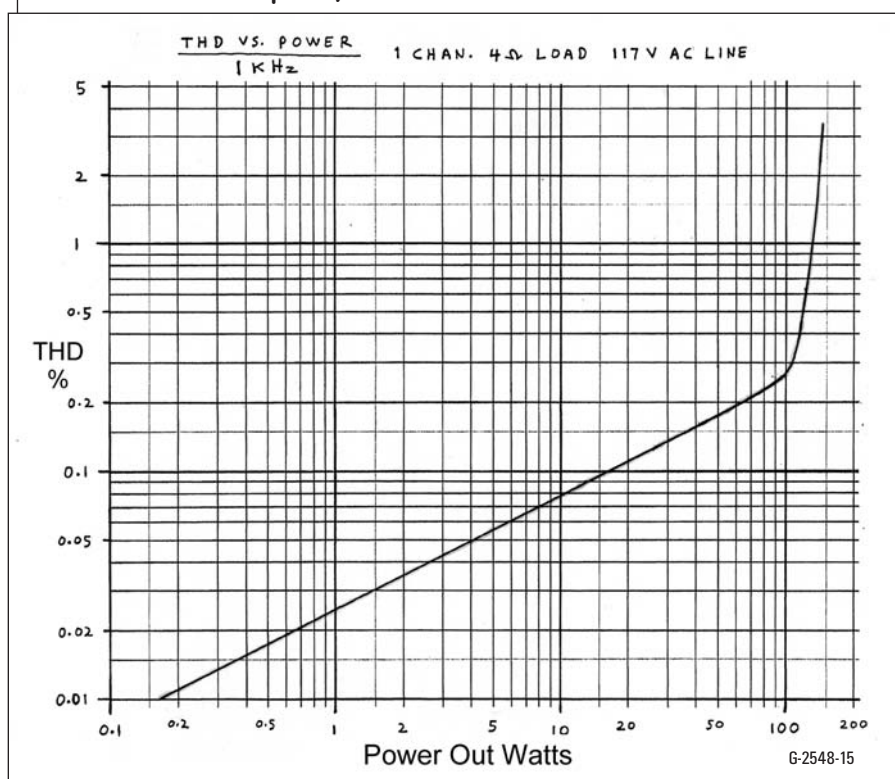
Figure 11 shows how positive current feedback (PCFB) generates a negative resistance in series with the positive output R of an amplifier. A unity-gain amplifier with two summed inputs (simplest example) has an output resistance of r_a . This drives a load r_L , which is ground-returned through a current-sensing resistor r_s , of low enough value relative to r_L to insignificantly affect the load current i_o . The analysis shows that the output resistance r_o , presented to the load, is $r_a - r_s C$. Thus r_o can be made positive, zero, or negative.

As adjusted in the Mad Katy amplifier, the plate-to-plate output resistance seen by the transformer primary has the following approximate values: (1) $2k\Omega$

with no FB, (2) 330Ω with the plate-fed

(yes, negative) with both FB loops. The 15dB conventional NFB, and (3) -150Ω PCFB loop has "added" a negative 480Ω

FIGURE 15: THD versus power, 1kHz.



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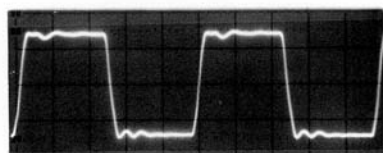
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resistance in series with the plate-to-plate output.

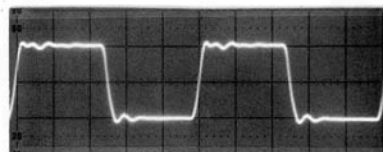
While the amp is normally adjusted for a DF of over 20 from 40Hz–10kHz, I had fun cranking up P4 (**Fig. 10**) fully, resulting in an output R of $\sim 8\Omega$ at the 4 Ω tap. Mathematically, the DF would be ~ 5 , but that does *not* mean “negative damping.” On the contrary, at a speaker’s bass impedance peak frequency the delivered voltage will actually *decrease*, lowering the “Q” of the bass rolloff. Mathematical “negative damping factor” equals electroacoustic *overdamping*.

What was fun was watching a sine wave from the amp output—first with no load, then seeing the voltage *increase* when connecting a 4 Ω load resistor! A mechanical analogy would be pushing a heavy wheelbarrow along level ground, then coming to an incline and finding it *easier* to push uphill—you might be “inclined” to instinctively look for a hidden motor!

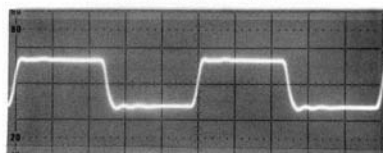
FIGURE 16: 10kHz squarewave responses.



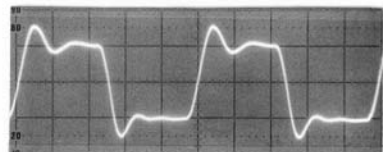
185W average 4 Ω load



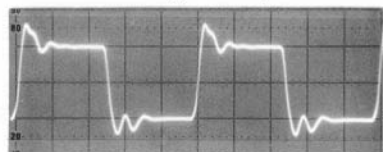
100W average 4 Ω load



10W average 4 Ω load



100W average 4 Ω /1 μ F load



±20V open load

G-2548-16

PARTS LIST CONTINUED

Ref.	Description			Mfr. P/N	Dist.	Dist. P/N
R26	Res, 15k, 5%	1/4W				
R6, 7	Res, 47k, 5%	1/4W				
R17	Res, 68k, 5%	1/4W				
R13, 14	Res, 100k, 5%	1/4W				
R24, 25	Res, 100k, 5%	2W				
R1, 9, 10	Res, 240k, 5%	1/2W	∇ (R1)			
R2	Res, 270k, 5%	1/4W	∇			
R19	Res, 330k, 5%	1/4W				
R4	Res, 360k, 5%	2W				
R8	Res, 470k, 5%	1/4W				
R3	Res, 1M, 5%	1/2W	∇			
R15, 16	Res, 1M, 5%	1/4W				
R5	Res, 1.5M, 5%	1/4W				
R11, 12	Res, 10M, 5%	1/4W				
T1	Audio Line transformer			Hammond 560C		
BR1	bridge rectifier				Digi-Key	36MB160A
BR2	bridge rectifier				Digi-Key	2KBB100
BR3	bridge rectifier				Digi-Key	KBC102
C6	cap, 10nF, 630V					
C5	cap, 1 μ F, 6.0V	poly	Solen		AES	C-FS1-630
C8	cap, 2.2 μ F, 630V	poly	Solen		AES	C-FS2D2-630
C7	cap, 10 μ F, 630V	poly	Solen		AES	C-FS10-630
C9, 10	cap, 47 μ F, 450V	electrolytic	Panasonic	EEV-EB2 W470	DK	P5951
C11	cap, 100 μ F, 350V	electrolytic	Panasonic	EEV-EB2V101	DK	P5938
C12	cap, 560 μ F, 10V	electrolytic	Panasonic	EEV-FC1A561	DK	P11186
C15, 16	cap, 1000 μ F, 15V	electrolytic	Panasonic	EEV-FC1C102	DK	P10253
C14	cap, 1500 μ F, 350V	electrolytic		ECE-T2VA152FA	DK	P10671
C13, 14	cap, 4700 μ F, 25V	electrolytic		EEV-FC1E472	DK	P10289
CL1-3	cap, 4700 μ F, 25V	electrolytic		CL70	Digi-Key	KC007L
D1, 2, 3, 11	diode, zener 10V			1N5240	Digi-Key	1N5240BDICT
D4-10, 12-14	diode, zener 68V			1N4760	Digi-Key	1N4760ADICT
D15	diode			1N4148		
F1	fuse, 10A SB				DK	F483
F2	fuse, 2A FB				DK	F119
F3	fuse, 0.25A SB				DK	F310
F4	fuse, 2A SB				Digi-Key	F325
LED1	LED, green				Digi-Key	67-1755
LED2	LED, red				Digi-Key	67-1611
LED3, 4	LED, blue				Digi-Key	67-1750
MOV1	transient absorber				Digi-Key	ERZ-V200201
PE1	Power entry module	PREO 10ERM1			Digi-Key	364-1042
Q1-3	power MOSFET	N channel	IRC	IRFBE30	Digi-Key	
Q4	power MOSFET	P channel	IRC	IRFR 9310	Digi-Key	
Q5	transistor	PNP	Panasonic	2SB1011	Digi-Key	
R41	Res, 0.33 Ω	35W	Ohmite		Digi-Key	TCH35PR330N
R24	Res, 100 Ω , 5%	2W BC			Digi-Key	BC100W-2CT
R37, 38	Res, 121 Ω , 1%	1/4W				
R13, 25	Res, 470 Ω , 5%	2W BC				BC470W-2CT
R35, 36	Res, 487 Ω , 1%	1/4W				
R42, 43	Res, 1k, 5%	1/4W				
R9, 10	Res, 2.7k, 5%	1/4W				
R30	Res, 5.6k, 5%	1/4W				
R31, 34	Res, 6.8k, 5%	1/4W				
R32	Res, 10k, 5%	1/4W				
R40	Res, 18k, 5%	1/4W				
R19, 39	Res, 22k, 5%	1/4W				
R33	Res, 27k, 5%	1/4W				
R5	Res, 27k, 5%	1W				
R29	Res, 33k, 5%	2W BC			DK	BC33KW-2CT
R17, 27, 28	Res, 47k, 5%	2W BC			DK	BC47KW-2CT
R7, 11, 15	Res, 47k, 5%	1/4W			DK	
R26	Res, 56k, 5%	2W BC			DK	BC56K-2CT
R14	Res, 68k, 5%	2W BC			DK	BC68K-2CT
R16, 22	Res, 100k, 5%	1W BC			DK	BC100KW-1CT
R14	Res, 100k, 5%	2W BC			DK	BC100KW-2CT
R21	Res, 180k, 5%	1W BC			DK	BC180KW-1CT
R12, 18	Res, 220k, 5%	2W BC			DK	BC220KW-2CT
R6, 8	Res, 470k, 5%	2W BC			DK	BC470KW-2CT
R20, 23	Res, 1M, 5%	1W			DK	BC1.0MW-1CT
S1	switch, DPST	20A NKK		S331	DK	360-1191W
S2	switch, SPDT	NKK		M2022S2A1W01	DK	360-1156
T1	transformer, 9V, 9A			Hammond 182N9		
T2	transformer, 350V, 1.8A			Plitron 097053201		
T3	transformer, 220V, 0.07A			Plitron 007029201		

This method of DF increase, if properly implemented, does not compromise stability or transient response, as opposed to using a large amount of conventional negative voltage FB. Also, because the current FB is already positive (with much less than unity loop gain), phase shifts can only steer the loop phase toward negative, which will only further increase the stability margin. Phase shifts, such as from LF and HF rolloffs, will reduce the DF increase. As set, the DF drops to 13 at 20Hz and 9 at 20kHz.

If you desire a relatively low DF, P4 can be set fully counterclockwise, shutting off the CFB loop. DF is then about 4.

P4 has a DF range (midband, where unassisted DF is highest) of approximately 4, through infinity (zero output R) to about -5. DF numbers are the inverse of a physical parameter (resistance), so the "infinity" spike is only a contrived mathematical singularity; as P4 is advanced, the amplifier's R_{OUT} smoothly decreases from (approximately 4Ω tap as an example) 1Ω, through zero, to -0.8Ω.

The only warning is that when set for a negative output R, the amp won't be short-circuit stable. With maximum P4 setting, the output R is about $-0.2 \times$ the transformer output tap value; the mag-

nitude of this is the lower load limit for stability, e.g., about 0.8Ω at the 4Ω tap.

Figure 12 shows the front and rear panels. **Figures 13-16** show some graphic performance data.

Next month we'll take a look at this unit's power supply and then begin construction. *aX*

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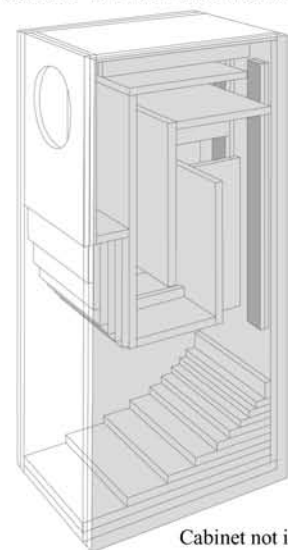


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PARTS LIST CONTINUED

Ref.	Description			Mfr.P/N	Dist.	Dist. P/N
T4	transformer, 9V, 1.67A			Hammond 182H9		
T5	transformer, 15V, 8A			Hammond 182N15		
VR1	reg. pos, adj			LM317T		
VR2	reg. neg, adj			LM337T		

Misc. (for both channels)

Description				Mfr.P/N	Dist.	Dist. P/N
Socket, octal (10)					AES	P-ST8-209M
Socket, 9 pin (4)					AES	P-ST9-211M
Fan, 12V DC 124mA	120 × 25mm	Papst		3412L	DK	381-1080
socket, octal (10)	1/16" hole 1 1/2" centers				AES	P-ST8-CIR-8
socket, 9 pin min. (4)	3/8" hole 1 1/8" centers				AES	P-ST9-511
fuse holder (6)	3AG	Littelfuse		342014	DK	F006
wire, green, 300V (100')	18AWG	Alpha		3055GR005	DK	A2064G-100
wire, red, 600V (100')	24AWG	Alpha		3070RD005	DK	A3070R-100
wire, blue, 600V (100')	24AWG	Alpha		3070BL005	DK	A3070L-100
wire, violet, 600V (100')	24AWG	Alpha		3070VI005	DK	A3070V-100
knob (2)	0.75" Ø	Kilo				226-4011
Sil-Pad K10 (20)		Berquist		K10-54	DK	BER114
Grommet (one 100 pkg)	for 3/8" hole	SPC			NWK	32F1353
Grommet (one 100 pkg)		SPC			NWK	31F2282
Handle (2)	4.562" centers	10-32 screws	HH smith		NWK	94F8976
Lug terminal strip (25)	2.625" centers	cinch		56A	NWK	29F591
wire, Teflon, 1000V, red (100')	22AWG	Alpha		5875 Red (3)	NWK	
rubber bumper (one 49 pkg)	0.70" Ø 0.38"	H SPC			NWK	92N4782
wire, yellow, 300V (100')	22AWG	Alpha		3051YL005	DK	A2016Y-100
wire, orange, 300V (100')	22AWG	Alpha		3051OR005	DK	A2016A-100
AC line cord	9' 10"	Qualtek		312040-01	DK	Q120
knob (2)	0.925" Ø	Kilo				226-4012



The Table Tuba

This author's latest horn design offers subwoofer capability, but also fits nicely in the living room.

By Bill Fitzmaurice

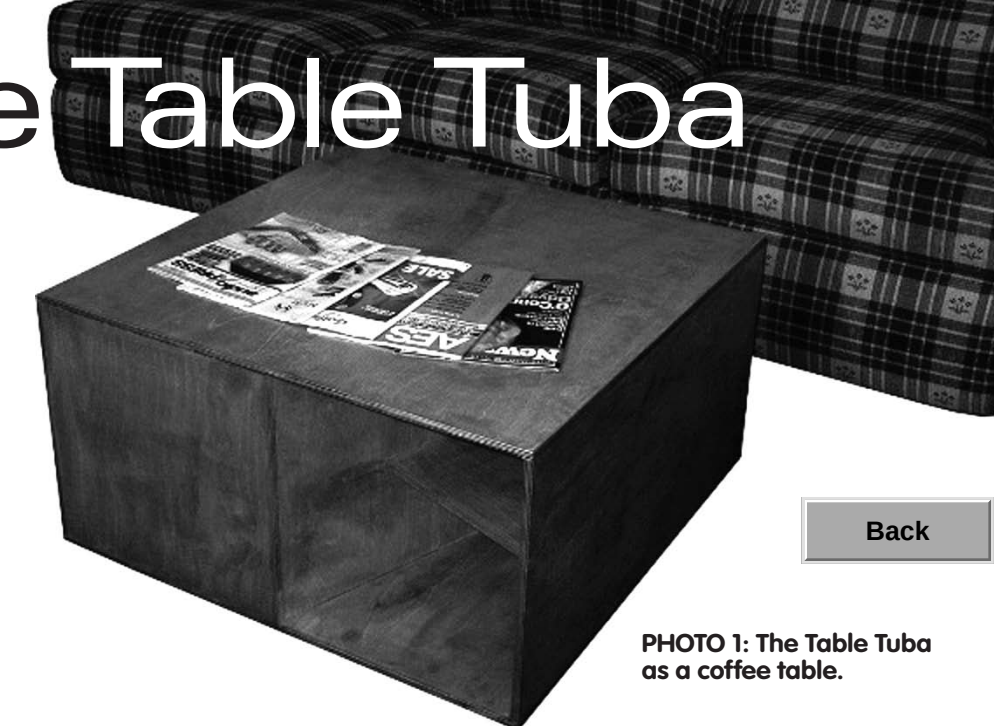
This project is in response to numerous requests for a Tuba folded horn for stereo and home theater capable of reaching down to 30Hz or lower. A 30Hz horn is easy; having space left for anything else in the room isn't. Then there's the matter of the WAF: Wife Acceptance Factor. The Table Tuba (Photo 1) gives you the sub that you want, cleverly disguised as a living room table to keep your wife happy. Or at least a little less unhappy when she demands to know why you need to have that "great big speaker thing" in her home.

FEATURES

This design is very driver tolerant. I tested it with a \$35 MCM 55-2421 eight (www.mcminone), as well as an Eminence HL10a ten that goes for four times that price, and there was very little difference between them. Target T/S specs are fs 30Hz, Qts 0.35, Xmax 8mm or more, Bl at least 12, Vas 30-60l. Most drivers with specs reasonably close to these will work fine in this box.

If you wish, you can widen the cabinet more than the original 15.75". I chose the dimensions so that there would be minimal waste when building from standard 5' x 5' Baltic birch plywood. Widening the cabinet will increase the mouth area for higher SPL and deeper extension, but not by much; you'd get only an additional 3dB by doubling it. I'd only widen it if I wanted to be able to fit dual drivers for higher power handling or lower impedance.

The throat impedance of this box is



Back

PHOTO 1: The Table Tuba as a coffee table.

very high and can nearly double the driver impedance (Fig. 1). If you want a 4Ω load, you must use a pair of 4Ω drivers parallel wired, or a dual 4Ω coil driver with the coils parallel wired. However, it is so efficient that you don't need to worry about maximizing your amp's output. An amp rated 100W into 4Ω will

put at least 50W into 8, and with the sensitivity of this speaker that's plenty for most home-theater setups (Fig. 2).

CONSTRUCTION

The entire cabinet is constructed from 1/2" material, either plywood, MDF, or particleboard. If you plan on a natural

FIGURE 1: Impedance chart.

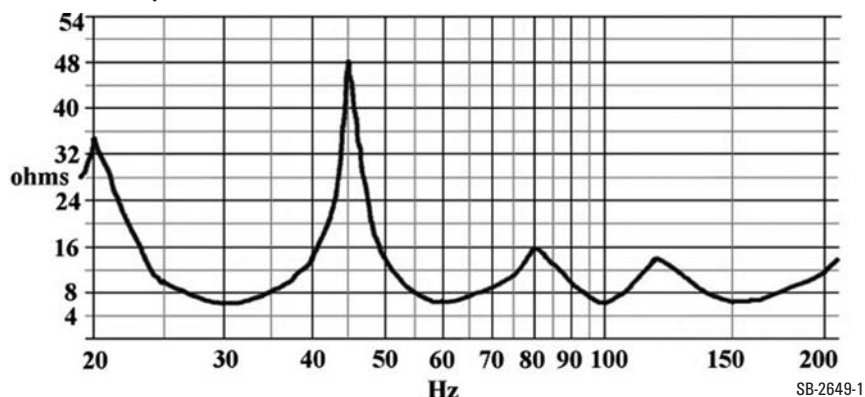
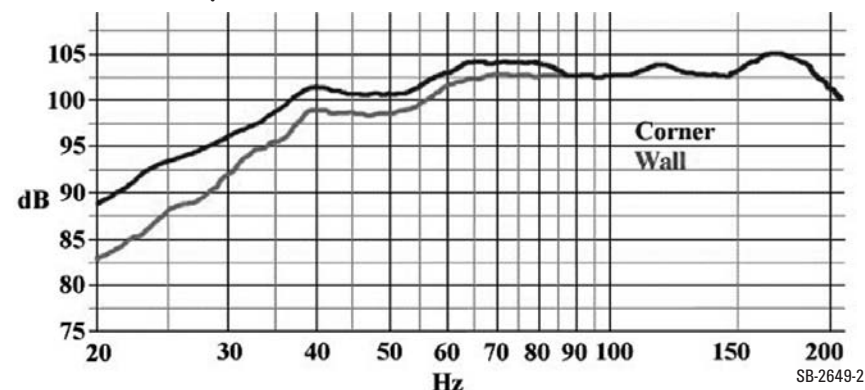


FIGURE 2: SPL response.



Because the joints in the cab must be airtight, I highly recommended that you use a polyurethane-based adhesive,

Construction starts by cutting out the two sides, both measuring 29.75" square. While the "sides" may technically end up being the top and bottom if you set the cabinet up as a low-boy table, I'll refer to the two largest pieces as the sides, and the cutaways as side views. Draw the locations of all the joints onto one side, as shown in **Fig. 3**. These dimensions show the actual clearances of the horn pathway, surface to surface.

Figure 4 shows the angles of the various intersections and the numerical

SB-2649-3

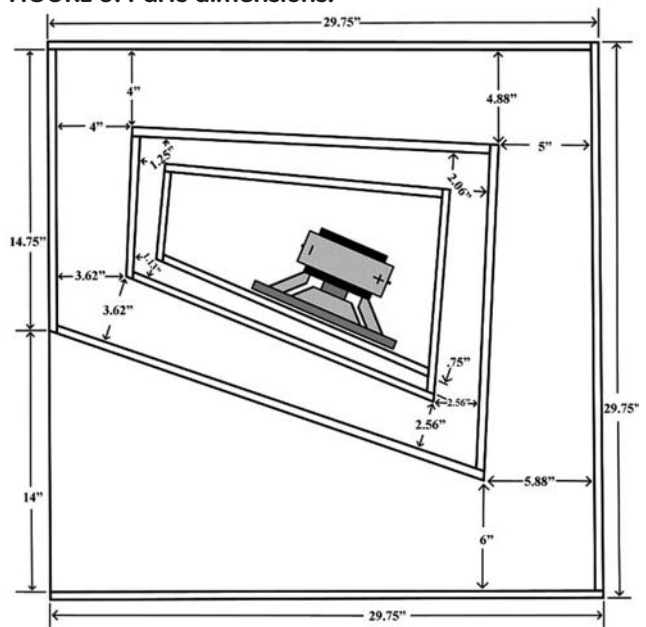
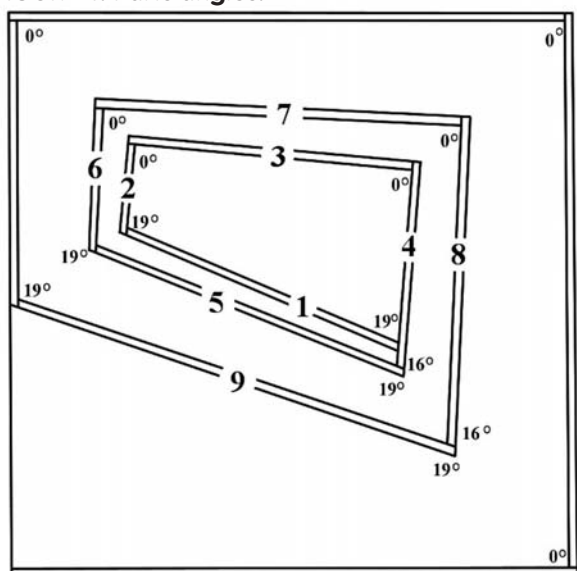


FIGURE 4: Parts angles.

SB-2649-4



designations of the internal panels. You don't need to utilize these angles in the layout, but notice that where the angle is 0° , such as at the junction of plates 7 and 8, the two parts are at right angles to each other. You will want to refer to these angles when cutting the panels, so write them on the side and they'll be there when you need them.

Figure 5 shows the nominal lengths of the internal panels. These are approximate dimensions only and depend on the actual thickness of the material you're building with. You must confirm all these sizes by measuring and/or dead reckoning during the construction process.

After you've drawn the joints on one side, put the two sides back to back and clamp them together. Drill pilot holes through the middle of each joint, using a drill bit the same diameter or slightly smaller than that of the nails you're using. Do not drill closer than 2" or so from the end of the joint, with the holes spaced 6–8" apart.

Drill the holes entirely through both sides, even if you plan on using screws instead of nails. Unclamp the two sides and put aside the second. Using either a circular saw or saber saw, cut out a driver access hole in the side as framed by panels 1–4.

The remaining parts are all cut to the same 14.75" width. If you're using a table saw, it's best to cut them all to width at the same time so that the rip fence can stay in the same place and all the parts

will be identically cut.

ASSEMBLY

Cut out panel 1, which is also the driver mounting baffle. Draw a line across the baffle $1\frac{1}{2}$ " from one end. Place the driver on the baffle, centered with one edge on the line. Mark the bolt locations through the frame holes. Put the driver aside. Drill the baffle for T-nuts for the driver mounting bolts. I suggest you use $\frac{3}{16} \times 32$ Allen head bolts, as the clearance inside the box is tight and you must

drive some of the bolts primarily by feel.

Draw two intersecting lines connecting opposing boltholes to determine the center of the driver and use a compass to draw the appropriate size hole on the baffle. Then cut this hole with a saber saw. Drive the T-nuts in place with a hammer, ensuring that they are located on the bottom of the baffle. Trial-fit the driver to the baffle to be sure that all the nuts are properly installed, then remove it and put it aside.

Attach the baffle to the side. The best

FIGURE 5: Approximate panel lengths.

SB-2649-5

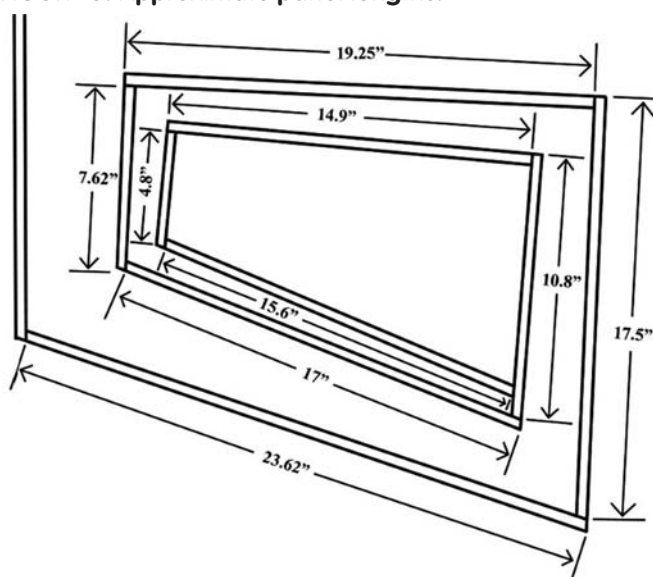
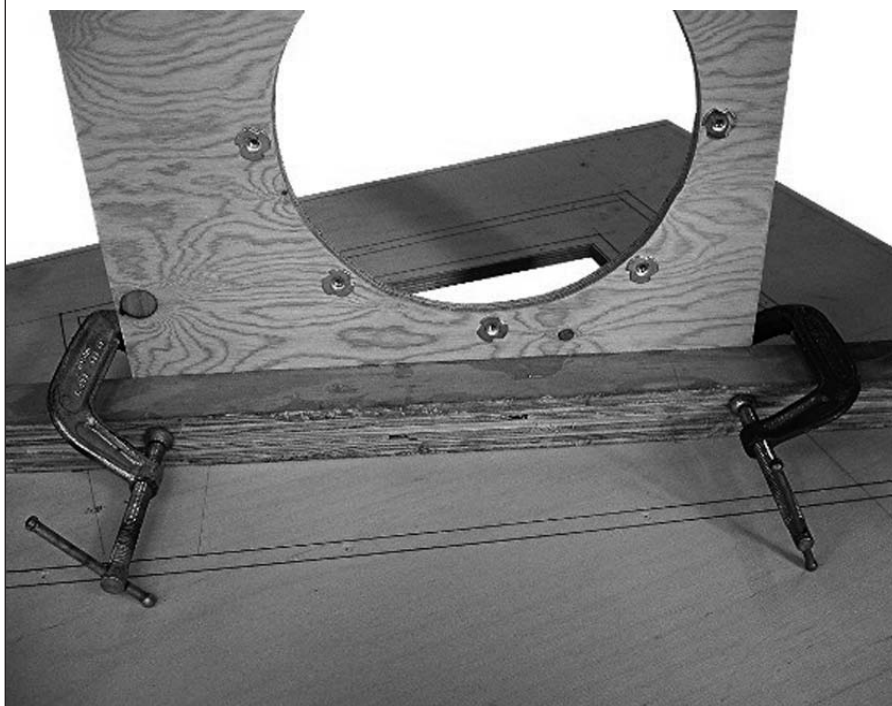


PHOTO 2: Attaching panel 1.



way to do this is to clamp a 2" x 2" guide board along the joint edge, apply adhesive to the joint, clamp the panel to the guideboard, and when you're sure all is right nail it in place (**Photo 2**).

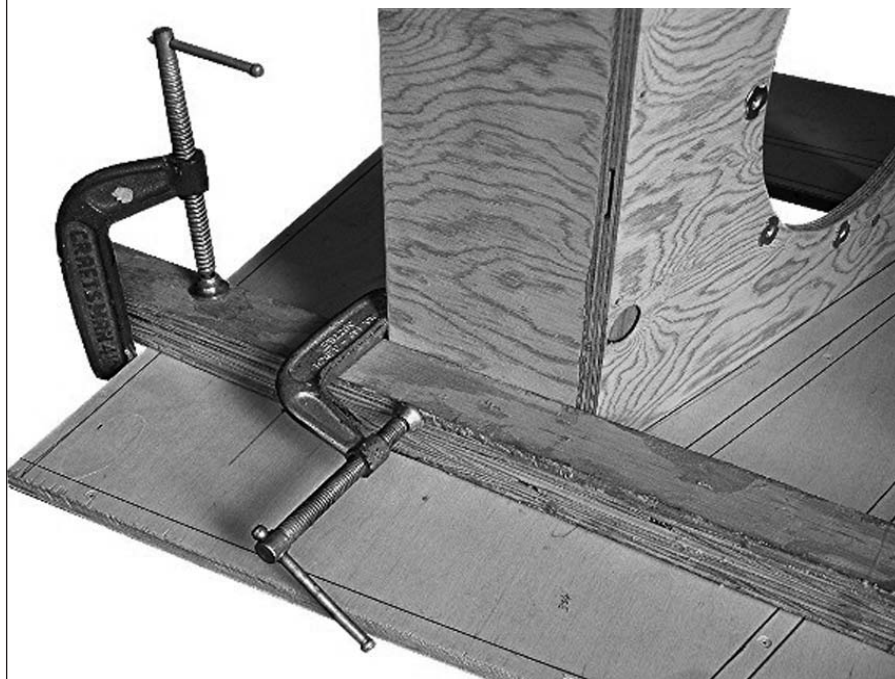
To prevent splitting, drill through the pre-drilled pilot holes about 1/2" or so into the panel. Use a nail set to drive the head of the nail below the face of the side. If you choose to use screws, follow

the same steps, using a pilot/countersink bit to prepare the hole before driving the screw.

Double-check the length and cut angles against the cabinet and cut out panel two. Attach it to the side using the same guideboard technique (**Photo 3**). Use a clamp at the end of the joint of the two panels and a scrap piece of plywood to hold the parts together while you first drill pilots and then drive two or three nails or screws along the joint, checking the alignment with a framing square to be sure that everything is square. Using the same technique continue with panels 3, 4, 5, and 6, checking alignment with a framing square as you add each new panel, holding things together with clamps until after you have driven the nails/screws (**Photo 4**).

Starting with panel 7, bracing is added between the panels (**Fig. 6**). Note that the braces don't touch each other and don't need to extend fully across the entire length of the panels. The braces are spaced midway across the cabinet. You figure their dimensions by dead reckoning (**Photo 5**). Place a piece of plywood

PHOTO 3: Attaching panel 2.



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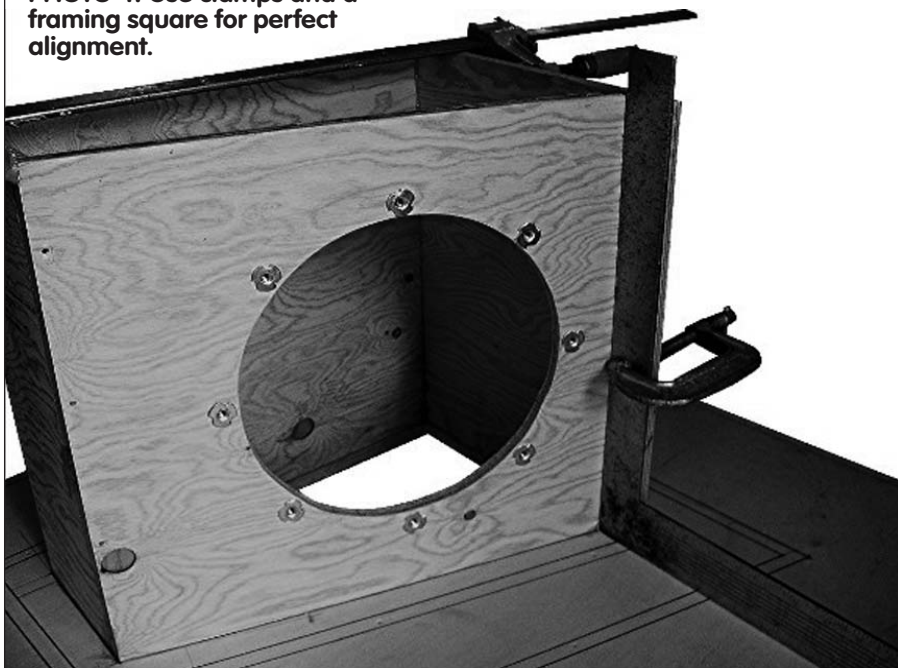


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PHOTO 4: Use clamps and a framing square for perfect alignment.



of sufficient size against the inner panel to which it attaches, in this case panel 3. Then set a straightedge across it where the next panel will joint to it, in this case panel 7, and draw the cut line.

After sawing, lay the brace in place again to check the accuracy of its sizing. Also draw on the panel where the installed brace will join to it. After cutting the brace that spans panels 3 and 7, install panel 7. Liberally apply adhesive to the edges of the brace and slide it into place, pushing it far enough to just be snug. Secure it in place with nails or screws driven through panel 7.

Continue the assembly process. When installing the braces joined to the front, top, and rear panels, use a framing square to check their alignment, pushing the brace in only as far as required to square the panels (**Photo 6**). When you install the cabinet bottom, do not install the brace between it and panel 9.

After installing the bottom, lay the cabinet on its side and apply a generous bead of adhesive on all the panel edges. Put the second side atop the assembly, making sure that it is oriented so that the pilot holes line up with the panels. Nail or screw it into place. If required, use a long pipe clamp to pull it into alignment with the rest of the cabinet.

After installing the side, go back and install the brace that goes between panel 9 and the bottom. Use a router with a pattern following bit or a sander to make flush the outer joints and those of the panels surrounding the access hole. Fill the holes over the nail or screw heads before sanding and finishing the box. After your finish is complete, install your driver.

SETUP

If you're using a ten, things are rather tight, so tight that reaching the far lower corner bolt isn't easy. Install that bolt first (**Photo 7**), with the driver offset so that you can reach it, and then swing the driver into place. Don't fully tighten any of the bolts until all are installed, then go around the frame easing them in until all are firmly set. This is a job made much easier with a small ratchet screwdriver.

Eight-inch drivers, on the other hand, have room to spare, even when mounted on a plywood reducer to fit over a

FIGURE 6: Braces (shaded).

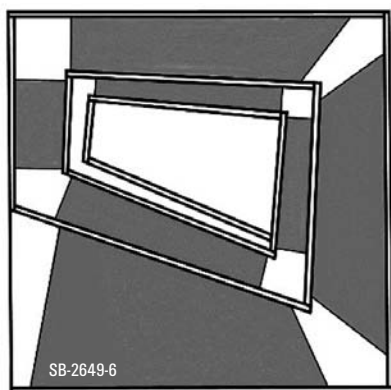


PHOTO 5: Dead-reckoning a brace.



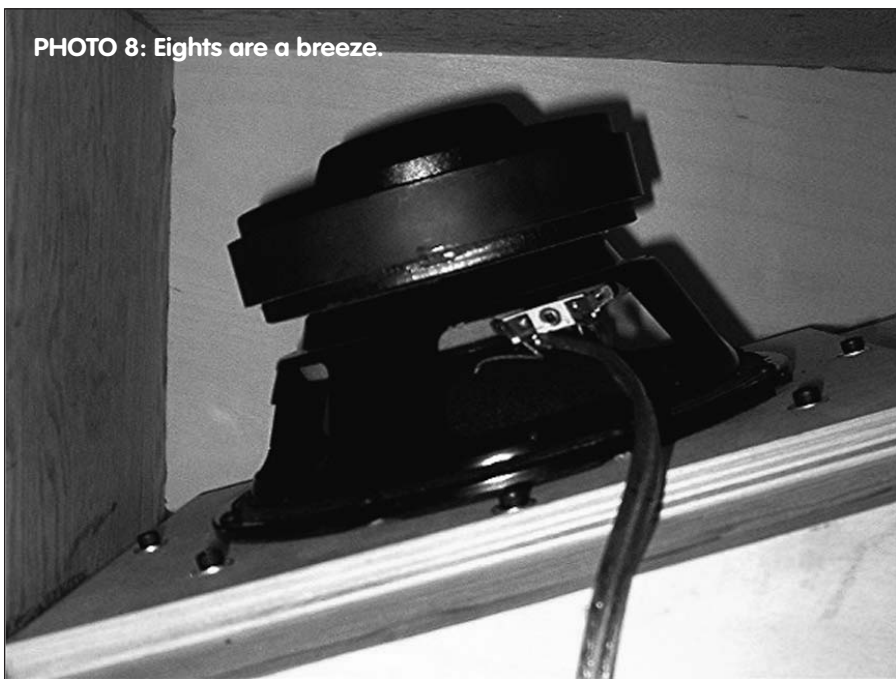
PHOTO 6: Keep it square.



PHOTO 7: Tens are a tight squeeze.



PHOTO 8: Eights are a breeze.



hole for a ten as I did in the prototype (Photo 8).

Finally, there is the driver access cover. Surprise! There isn't any. Just wire up your driver and plug it in.

Most bass horns have a sealed rear chamber that is employed to both tune the cabinet response and apply restorative force to limit the excursion of the driver. However, if the horn is long enough, and has sufficient throat impedance, the air mass of the horn alone will tune the driver response and limit excursion. That is the case with

the Table Tuba, and as a bonus the rear wave of the driver also adds to the low-frequency output of the cabinet in much the same fashion as a bass reflex cabinet. Computer simulation predictions show the excursion of the driver at 15Hz unaffected by rear chamber size, while at 10Hz the excursion differential between a 10-liter chamber and no chamber is only 10%.

As to placement, like all subs the Table Tuba (TT) benefits greatly from placement near walls or room corners, though you will still get very good re-

sults without corner loading. With upright wall placement (Photo 9) put the horn mouth at the floor, the access hole facing the wall, the cabinet 1" away from the wall. With corner placement the mouth is at the floor facing into the corner, the end of the cabinet 6–10" from the corner, the access hole again facing the wall from 1" away.

You may also place the TT in the corner on the floor, with the mouth in the corner and the box on 1" legs. If you use middle of the room placement, also use 1" legs, so that the rear wave is not blocked.



PHOTO 9: Upright for wall or corner placement.

Using more than one cabinet will enhance sensitivity via mutual coupling, but there is a finite limit. In the case of the Table Tuba, that limit is about 106dB/watt. As you can see in Fig. 6, a single cab reaches that limit only at 160Hz. With every doubling of the sum total mouth size, another 3dB of sensitivity is gained, but only where that 106dB limit has not already been achieved. In theory enough cabinets would allow 106dB sensitivity right down to the horn flare frequency.

Because the length of the horn in the Table Tuba is some 13', 106dB sensitivity could extend to 20Hz. The bad news is that it would take perhaps 32 Table Tubas to do so. No matter. Furniture or not, I can guarantee that my wife will draw the line at two. *ax*

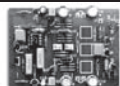


PHOTO 1: The modified HP test unit.



A First-Rate Audio Test Instrument, Part 1

A high-quality sine-wave generator, AC RMS voltmeter, and a worthy candidate for morphing into a reasonably high-power audio dummy load—all in one compact carry case and for a reasonable price!

By Todd Sharp

Recently a friend of mine asked me to put together a basic audio test bench. Other than scouting out a decent oscilloscope and DVM, I decided on an excellent candidate for modification: the HP 3550b transmission test set. The 3550b, with the following modifications, makes a downright sexy addition to any audio test bench. This endeavor proved so useful that I thought it worth sharing with *aX* readers (Photo 1).

FEATURES

If you have been to an electronics swap meet or surplus store in the past 20 years, I guarantee you have tripped over at least one of these sets. They also tend

to be readily available on eBay, where the prices vary, from as little as \$25 to as much as \$150. On eBay search for "HP 204c," "HP 403b," or "HP 353a" rather than HP3550b.

The individual model numbers are those pieces included in the set, along with a sturdy steel case with IEC AC service connector, removable cover, and shoulder strap. These were obviously used for field testing of transmission lines of some sort, though with a little ingenuity you can transform yours into a first-class audio test set. They often come with a nice set of patch cables, to boot (Photo 2).

The 403b AC RMS voltmeter (Photo 3) represents typical HP quality in a

PHOTO 2: The HP3550b before modification.

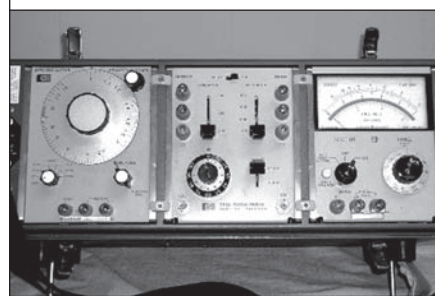
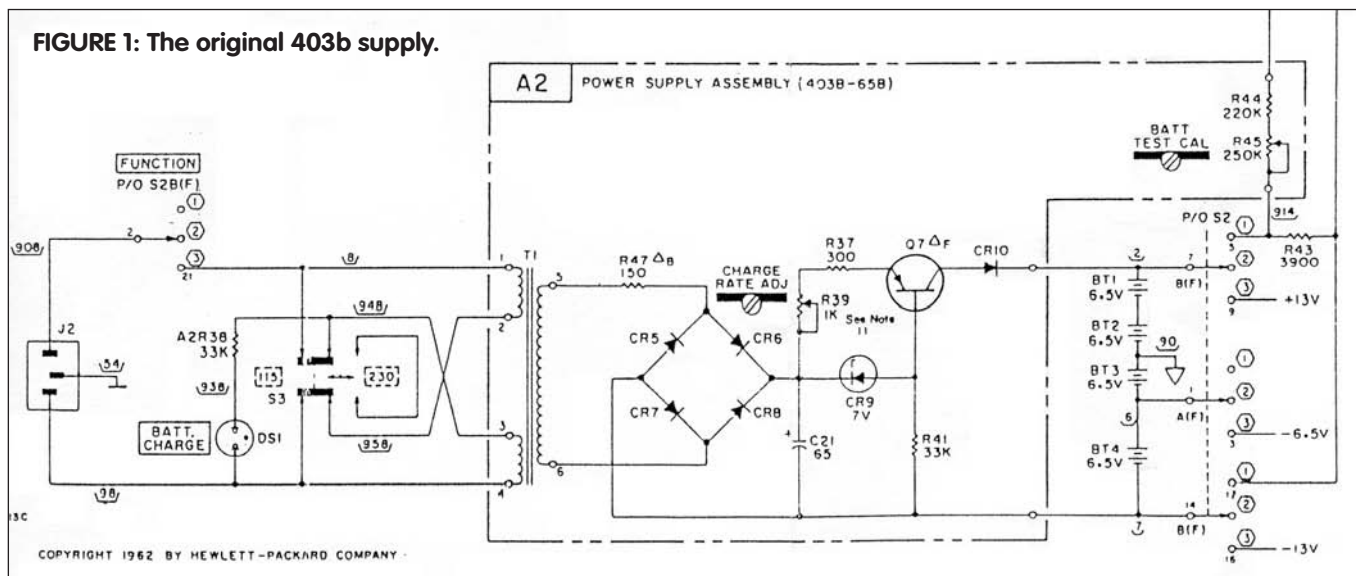


PHOTO 3: 403b voltmeter (front).



FIGURE 1: The original 403b supply.



compact package with a large viewable meter face. The full-scale meter range is from 0.001V AC RMS to 300V AC RMS. Input impedance is 2M Ω , and the bandwidth is from 5Hz to 2MHz. Frequency response is extremely flat (0.2dB 10Hz to 1MHz) through the audio band, dropping off to -3dB at 2MHz.

The scale is graduated in typical analog VM fashion in 0-3, 0-10, and dB (ref. 1mW into 600 Ω) scales. The dB scale is particularly handy in tracking frequency-response relative measurements. I have probably repaired about a half-dozen of these units and have found them to be easy to service and calibrate. Two of them have been in daily service at our shop for nearly eight years.

The one problem that they all seem to suffer in is the power supply, but fixing this is usually all there is to resurrecting them. They were originally fitted with 6V nicad cells and a charging supply. The nicad cells were configured as 6V ("C" cell size) batteries. I would guess that these are all at least 15+ years old at this point and the nicads are shot. I have tried rejuvenating them, but this is pretty much a lost cause, and finding 6V replacements has proven nearly impossible.

The idea of using batteries to operate gear like this is a good one, because battery-operated test equipment offers a few fringe benefits such as ground isolation, virtually ripple free supply, and otherwise unaffected by line noise and/or potential differences between chassis. But, you can

easily resurrect them for AC line operation by simply replacing the batteries with electrolytics and a few zener diodes for regulation. Add a few resistors and voila—you are in business. Both the 403b and the 204c use the same 6V nicads and almost the same supply configuration.

To save space in this article I will show only the power-supply schematics that the proposed modifications deal with. I will upload more complete info from the HP manuals on my website (www.amprepair.com) for those who might need or want it.

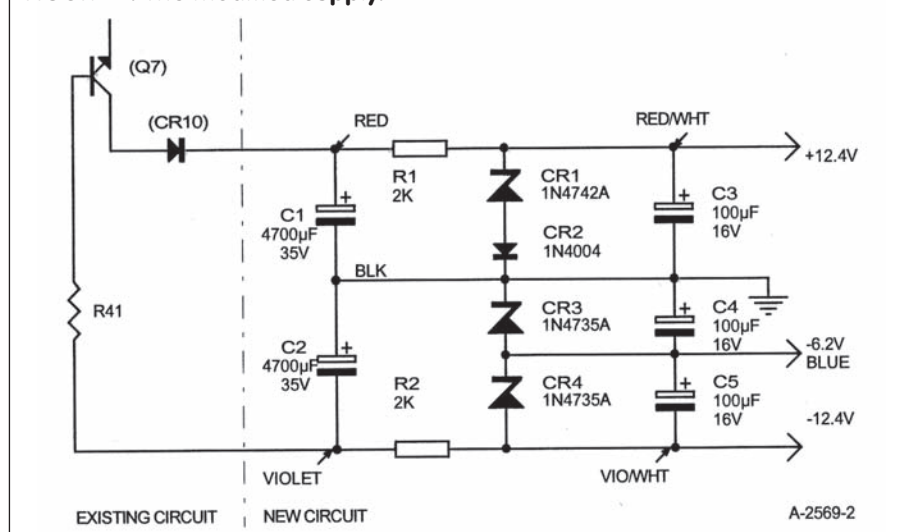
403B MODIFICATION

This mod is dead simple. You have a +13V DC supply and a -6.2V DC and a -13V DC supply, all of which you can

easily achieve with a tapped series string of zener diodes. I suggest that you leave all of the chassis components in place (**Fig. 1**), even though some of them are no longer needed because you are no longer charging batteries. They don't hurt anything, and leaving them in place really makes this a simple modification.

I must confess that I did not go to the extent that most *aX* authors usually do with massive filtering, bypassing, and other such "POOGEing." I just made a nice power supply that works well without having to design PC boards and so on. Go crazy if you prefer with bigger caps and film bypass if you think it will matter—and if you can fit it all in. The modified supply circuit is shown in **Fig. 2**. The new circuit components will re-

FIGURE 2: The modified supply.



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place the nicads.

Remove the left side and top panels (**Photo 4**). You might as well plug it in and see whether it works at this point. Locate the batteries and the red, violet, blue, and black wires attached to the batteries. The black (or perhaps white with black stripe) is ground. Red is +13V, violet is -13V, and blue is -6.5V.

See whether these voltages are present. If you are lucky enough to have something close to a power supply, then go ahead and see whether the meter responds properly to input. Usually the supplies are barely there at all—hence the reason for the mod.

Make a drawing/legend of the push-pin connections to the top board. Remove the push-on connections and re-

move the top board. Remove the left side frame. The batteries should now be exposed. Clip the (red, violet, blue, black) wires from the Nicad cells and remove the batteries. Clean up any leaked battery goo (**Photo 5**).

There were various-size cells used in these units, though I always seem to find a way to adapt the battery mounting apparatus to mount up the filter caps and secure them to the chassis. Arrange the two 4700 μ F/35V caps to the chassis so that they are joined at the center (ground) with a positive lead from one and the negative lead from the other. You can then simply tack-solder the remaining components to these cap leads (**Photo 6**).

Re-attach the wires you removed from

the batteries as follows:

Red to the positive input terminal of the 4700 μ F positive filter. Violet to the negative input of the 4700 μ F negative filter. Black to the joined ground node, and blue to the node where CR3 and CR4 are joined.

Clip off the red/white and violet/white wires from the front panel three-position power switch and attach them instead to the zener-regulated nodes—red/white to the cathode of CR1, violet/white to the anode of CR4 as indicated on the modified schematic (**Photo 7**, **Fig. 2**).

Remove the bottom cover and locate the AC inlet terminal, which has only one wire coming off it. Break this connection and insert the 1/8A fuse in series with this line, making sure to insulate with shrink tubing. That should be all there is to it.

Plug in the power cord and check with a DC voltmeter that all three supplies (+12.4V, -6.2V, and -12.4V) are there. If the supplies are good, you should have a functioning AC RMS voltmeter. Visit www.amprepair.com for links to further calibration information.

The specs for the **204c sine-wave generator** (**Photo 8**) are as follows: (modified unit) maximum output is 5.4V RMS unloaded, 5.4V RMS into a 10k load, and 2.7V RMS into 600 Ω . THD is 0.08% (1kHz at full output). Frequency response is virtually flat throughout the audio band. It does become a little wobbly at line frequency (60Hz), but this 60Hz beating will nearly disappear if you lift the AC ground. Flatness is otherwise within 0.2dB from 5Hz to 200kHz, -1dB at 500kHz, and -3dB at 1MHz.

POWER-SUPPLY MODIFICATIONS

You have almost the same supply as the 403b (**Fig. 3**). You need +12V and -12V to achieve a 5V RMS output. I opted to go with the same basic design utilizing a simple zener-regulated supply. Again, feel free to beef up the caps, although the specified parts used here work just fine.

Though the 204c has an off position on its range switch, the stock supply is “on” as long as the AC is plugged in (unlike the 403b) due to the original charging circuit. I don’t like the idea of leaving things on when I leave the bench, so I will add a

PHOTO 4: Uncovering the 403.

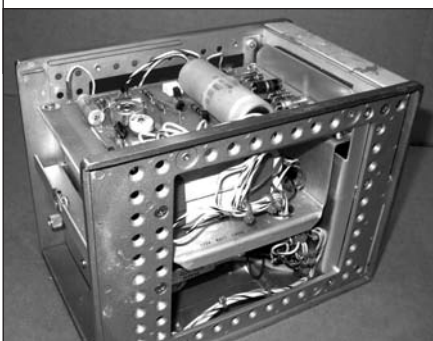


PHOTO 5: Exposing 403 innards.

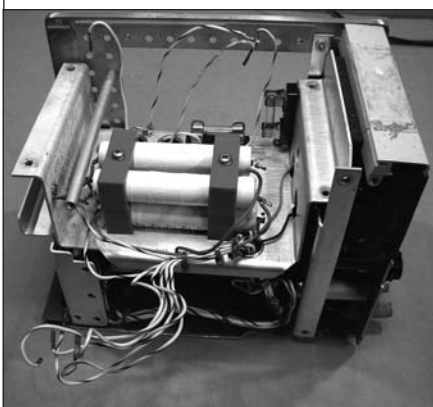


PHOTO 6: 403 wiring.

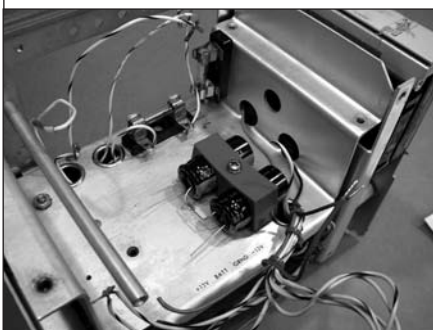


PHOTO 7: Re-attaching the wiring.

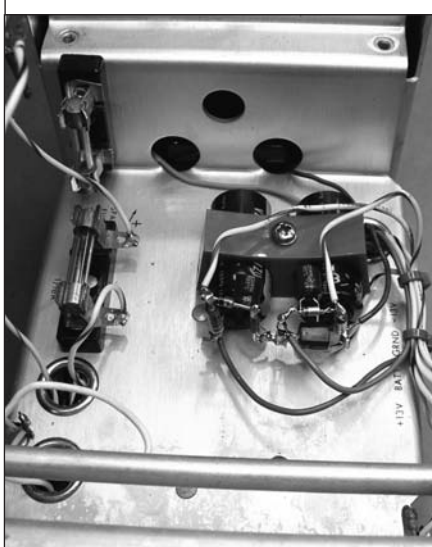
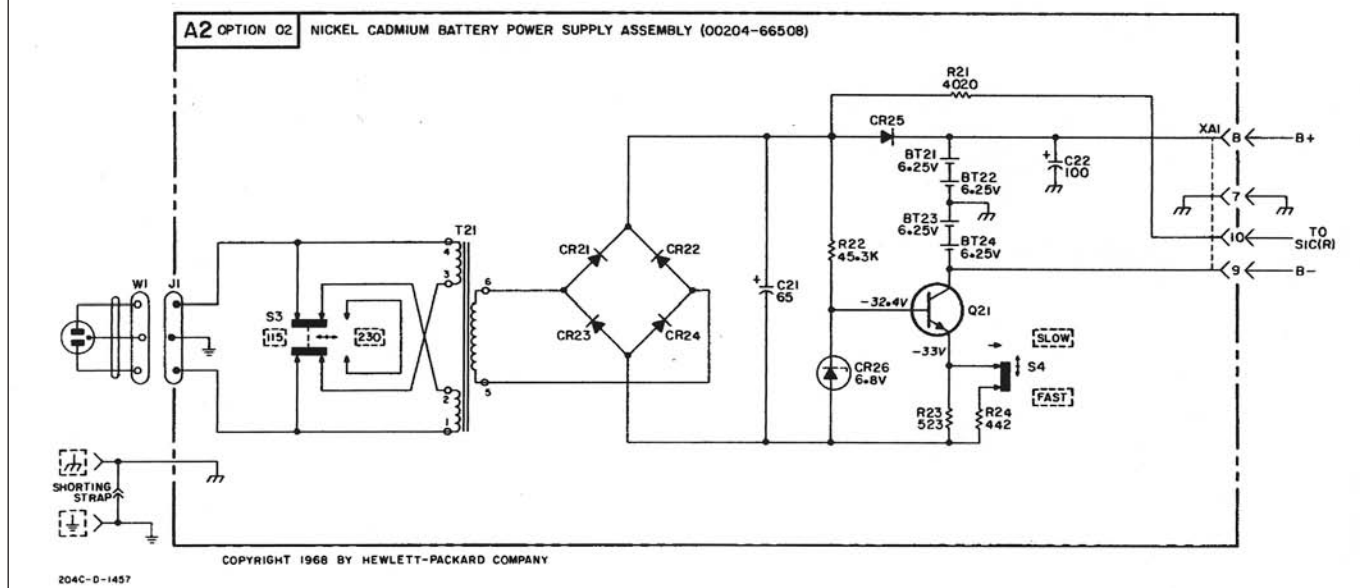


PHOTO 8: 204c sine-wave generator.



FIGURE 3: Original 204c power supply.



power switch to the front panel as well as a line fuse.

Note the original power supply and the modified supply. Remove the nicads and find plenty of space to replace them with your new components. Again, you may need to do a little cleaning on the nicads, which often leave leakage residue on the board.

Remove the top, bottom, and both side panels. Plug the unit in and power it up to see whether you have a supply. There are test posts available at the side marked B+ and B-. If your batteries are in reasonable shape, you may see +12 and -12V DC with reference to ground. More than likely the supply is not functioning properly due to defective nicads. The power-supply module is attached with four screws through the side frame brackets. Remove these screws to pull the supply portion away from the back of the oscillator (**Photo 9**).

Disconnect and remove the four nicad cells. If they are mounted in with friction clips, unsolder and remove these too, so you can use their PC throughholes for the new components. Notice that there is a 10-contact bifurcated plug at the card edge that interfaces to the main part of the unit. Use contacts #2 and #4 to run one side of the AC line up to the newly added power switch. In order to do this, you will need to remove any connections at these two places—on both the PS card and its mate connection traces on the main unit board.

There are different versions of the 204c out there, so I am not sure that yours will exactly match the specimen in this article,

but I have yet to open one that did not have the rechargeable battery option. In any case, you will need to make sure that contacts #2 and #4 are “free” if you are to utilize them for your power switch AC line interrupt.

There is a PC trace connecting a rear panel ground post marked with a flat line

(as opposed to the one next to it, which is marked with the ground “arrow” symbol). This rear post connects to the #4 contact at the rear of the PC board connector via a small jumper. Remove this jumper and then open up this trace where it connects to the chassis. I use a Dremel with a little router bit for this, which works like

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a charm, though a good sharp knife along with a scraping edge of some sort will do just fine. See lower left corner of **Photo 10**.

Interrupt one side of the AC connector where it goes to the voltage selector. It is the gray wire in **Photo 10**. Attach a wire from this side of your AC inlet to the PC pad where you removed the jumper from the rear panel ground post. Verify with

an ohmmeter that this pad is in fact connected to the #4 contact and also that it is no longer at chassis potential. Be sure to run this and all other wires under the PC board so they will not be crimped when you re-install the power module on final assembly.

Next attach a wire to the #2 contact underneath the board. This is currently an

unused contact, so you will just carefully tack on to it. Just to be sure—verify that this is not attached to anything before proceeding. The other end of this wire that you tacked to #2 goes to the line selector switch. This would be what you disconnected from the AC inlet in the last step.

Now you should look at the main oscillator unit card edge #2 and #4. Remove

PHOTO 9: Disassembling the 204c.

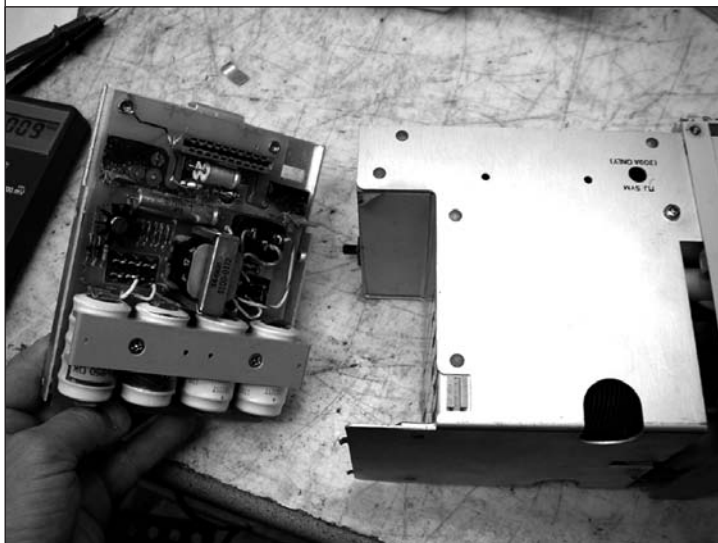
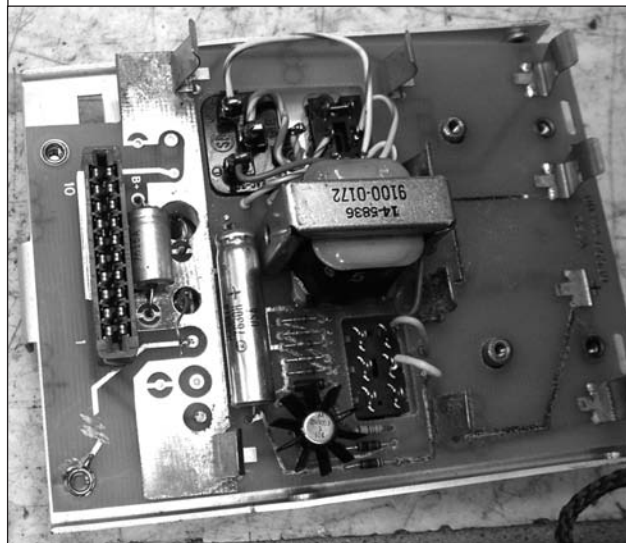


PHOTO 10: Exposing 204 innards.



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PHOTO 11: Uncovering the 204.

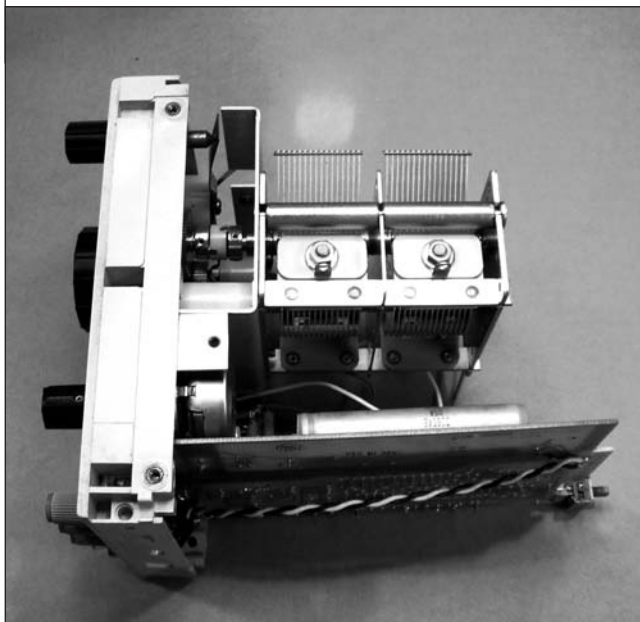


PHOTO 12: Notch in cover accommodates power switch.

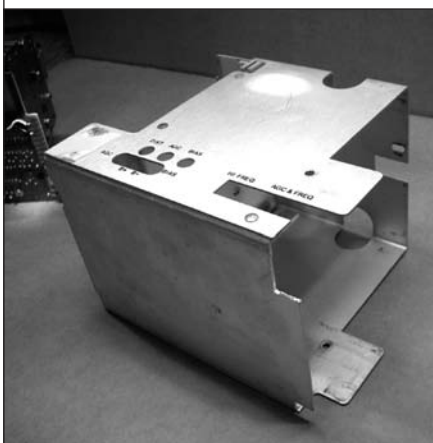
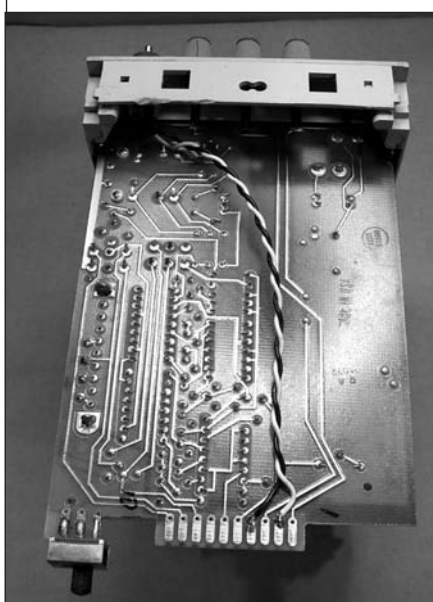


PHOTO 13: New 204 connection.



both side panels and then one screw in each side of the cover and the one nylon screw at the top. Pull the cover away from the oscillator unit (**Photos 11 and 12**).

Now that both sides of the main board are exposed, remove any connections to #2 and #4 card edge and affix a twisted pair as shown in **Photo 13**. There is a spot to install your AC switch—just below the range switch. Use shrink tubing to insulate these connections.

Note that you will need to cut a slight notch in the cover. You can see this right about in the middle of **Photo 12**. This will make room for the newly installed power switch.

REMAINING MODS

Remove Q21, CR26, R22, R23, R24,

and jumper across CR25 (see **Fig. 3**). Once you have done that, you will need to install a jumper between the negative FWBR output (the negative end of C21) and where the collector of Q21 used to be (the input side of R4, **Fig. 4**).

As you can see, you are simply replacing the batteries with filters, current-limiting resistors, and 12V zeners. You will also need to install a quasi center tap at the secondary of the power transformer (T21) with a pair of 1k, 2W resistors as shown. You can wick out the PC traces and pads where the battery clips used to be and use these pads for your new components.

You will also need to interrupt the existing PC trace connections to the card edge connector #8 (+12V) and #9 (-12V), so you can insert the series resistors (R3 and R4). This will require that you run a wire from the output of the +12 rail to pin #8 and another wire from the output of the -12 rail to pin #9. Again, run these wires under the PC board. The ground is already intact at pin #7 (**Photo 14**).

I used a dab of glue underneath the big electrolytics to hold them in place

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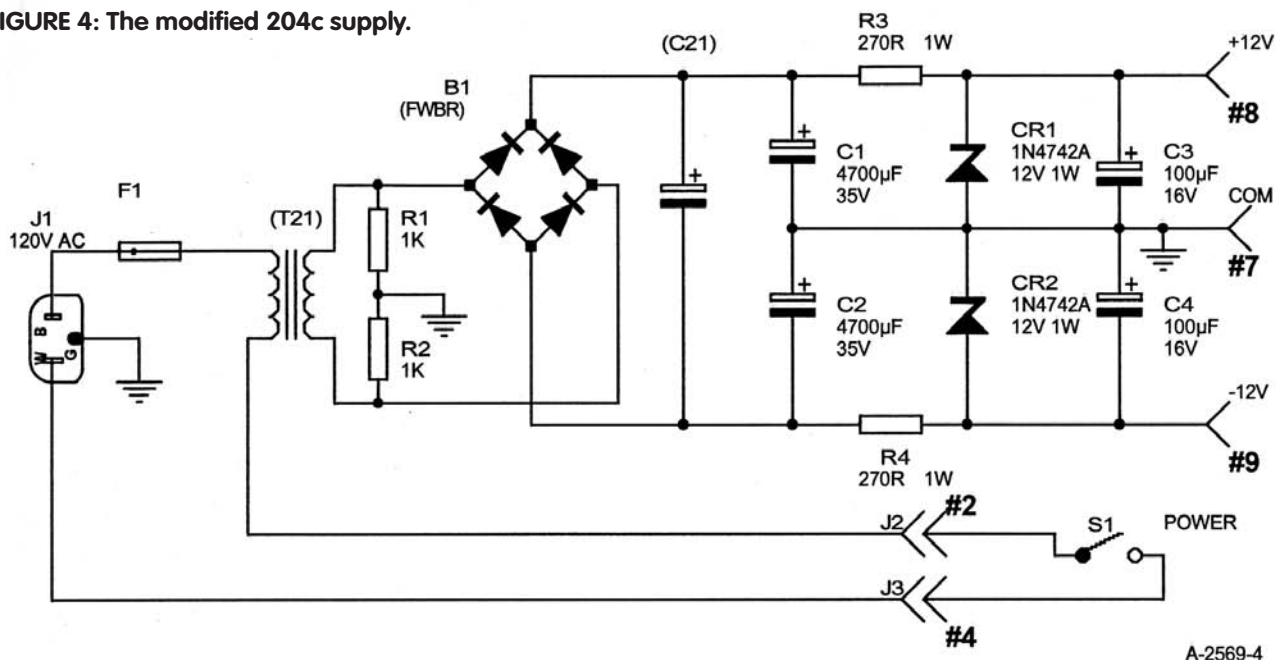
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FIGURE 4: The modified 204c supply.



A-2569-4

and wired the 100µF caps directly across the zeners, which is why they are somewhat obscured in the photo. Finally, replace the wire connecting the other side of the AC inlet to the voltage selector switch with the 1/8A fuse. I think everything ought to have a fuse.

After checking all your connections and polarities, temporarily jumper the switched side of the AC line straight to the voltage selector and fire up the supply. I have the luxury of using a Variac with current metering to gently power up my work. This is very handy and often

reveals the bonehead mistake before . . . smoke. So if you are just plugging in, then check your connections again and let 'er rip.

You should have +12 and -12V at pins 8 and 9 with respect to pin 7. Check to see that you have about ±20V

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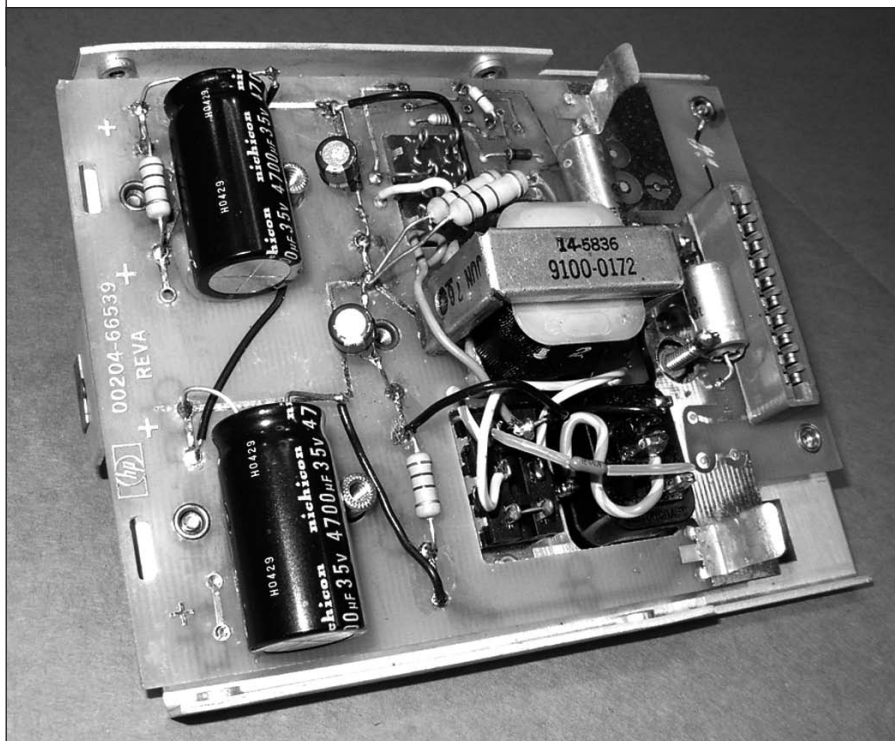
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PHOTO 14: Re-wiring the 204c.



at the inputs of each 4700µF filter. If all looks well, tighten down the screws that attach the PC board to the rear panel (make sure nothing is crimped under there) and plug it into the oscillator. Turn it on and check your $\pm 12V$ again at the test points at the side of the oscillator.

If all is well, you should have a nice-looking sine wave at the output. The front panel middle post is ground. If your oscillator needs calibrating, you may find links at my website: www.amprepair.com. It's very simple, requiring only a scope, DVM, and a frequency

counter. Next month, in Part 2, we will go to work on the HP 353a unit. *aX*

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Mouser

www.mouser.com

800-346-6873

Digi-key

www.digikey.com

800-344-4539

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403b Power-Supply Mod Parts

Component	Description	SOURCE
CR1	12V 1W (1N4742A) zener diode	Mouser # 78-1N4742A
CR2	1N4004 diode	Mouser # 512-1N4004
CR3, CR4	6.2 1W (1N4735A) zener diode	Mouser # 78-1N4735A
C1, C2	4700µF/35V radial electrolytic cap	Mouser # 647-UVZ1V472MHD
C3, C4, C5	100µF/16V electrolytic cap	Mouser # 647-UVZ1C101MDD
R1, R2	2k ½W resistor 5%	Digi-key # 2.0KH-ND
F1	½A micro fuse	Mouser # 504-MCRW-125MA

204C Power-Supply Mod Parts

C1, C2	4700µF/35V radial electrolytic cap	Mouser # 647-UVZ1V472MHD
C3, C4	100µF/16V electrolytic cap	Mouser # 647-UVZ1C101MDD
R1, R2	1K 2W FP metal oxide resistor	Mouser # 282-1.0K
R3, R4	270R 1W CF resistor	Mouser # 294-270
CR1, CR2	12V 1W (1N4742A) zener diode	Mouser # 78-1N4742A
Sw1	power switch Alco SPDT	Mouser # 506-MTA106D
F1	½A micro fuse	Mouser # 504-MCRW-125MA
Misc	power switch protective guard	Allied Electronics # 948-0002

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Servo Subwoofer Design Revisited

This father and son team up to show us the ins and outs of designing and building a servo subwoofer.

By Art and Jac Brown

Art: I started building speakers in 1950, when I was trying to develop a back-loading horn, using a 12" driver and the room walls for horn extension. My friends were also trying to build speakers, with one concentrating on an infinite baffle of about 25ft³, another a ported speaker, and a third, a tuned pipe enclosure. The friend who worked with the ported speaker also invested in the best driver, costing about \$35. His system sounded best when compared with the others at my home. Our source of information was *Audio* magazine and some of the references it contained.

During the next few years, I became interested in the acoustic suspension speaker system. I was unable to put much money into drivers (\$12 for a 12" diameter coaxial speaker), so I modified the cone suspension of my existing driver trying foam and chamois as surround materials. I also increased the voice coil travel by rewinding the coil. Some improvement was evident. I had panel vibration out of this world, though, but it was well damped when I had my wife sit on the offending panel, which is *not* a practical solution. I finally found funds in 1960 to buy an AR-1 speaker, which satisfied me for a number of years.

Of course, in the 60s we were listening to monaural sound. I stayed with

monaural into the 80s when my son wanted to buy a stereo and took me along to several listening rooms. The system he bought sounded so good that I started having ideas about stereo for myself.

By the mid-80s, my son had grown up listening to my tales of speaker building, so when a co-worker told him about *Speaker Builder* magazine, we became interested. We read a number of issues and I decided to build a two satellites/subwoofer system based on what I learned from *Speaker Builder*. I continued to use the AR-1 as the subwoofer. In the January 1989 issue of *Speaker Builder*, I described my satellites.

By late 1990, I had completed developing a servo subwoofer, which I described in *Speaker Builder* 1/91. I concluded that the subwoofer's "distortion, staging, and transient handling are unmatched by any system to which I have listened." My son decided to build his version of a stereo speaker system with a servo subwoofer. This article discusses the development of the subwoofer.

Jac: Why build a servo subwoofer? Because the bass is so amazingly clean. Over the years, I've built various passive woofer systems, some of which had deep, strong bass, but there has always been a little bit

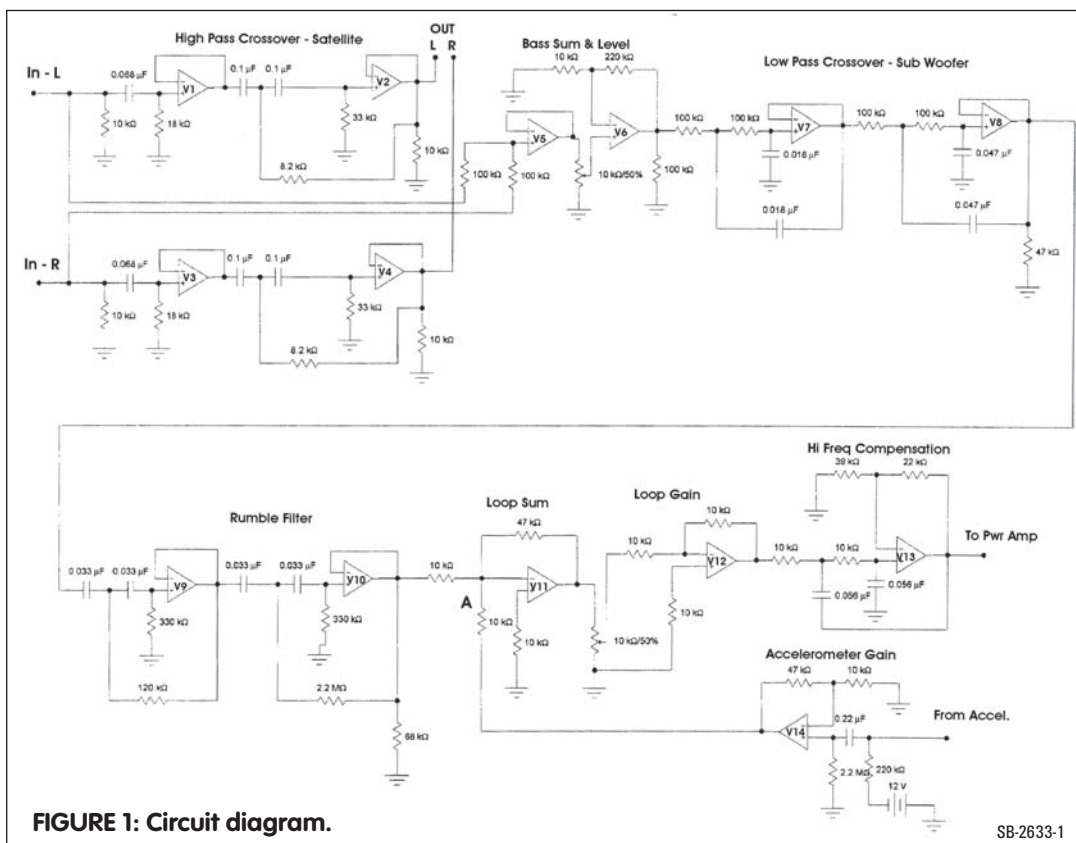


FIGURE 1: Circuit diagram.

SB-2633-1

of bass distortion. My father's servo subwoofer always seemed so clean that I just had to build one.

SOFTWARE TOOLS

Art: One of the big differences between my project and Jac's was the software and hardware available for measuring and simulating speakers. I used to need 8 hours to characterize a driver, and now with the computer it takes only a few minutes.

Jac: I used LAUD for speaker measurements, Electronics Workbench for circuit design, and an old version of SoundEasy to shape the filters including the speaker response and phase.

OVERALL SYSTEM

Art: This speaker system consists of the two satellites and the subwoofer along with the electronics for crossover and closed loop. The system gets its input from the receiver, which controls the CD player, DVD, and TV. The receiver pre-amp outputs connect to the servo amplifier board that has an electronic crossover between the subwoofer and satellites. After passing through the high-pass crossover, satellite signals are returned to the receiver, amplified by the receiver power amps, and passed to the satellites.

The signal for the subwoofer is the sum of the left and right channel that is processed through the low-pass filter (crossover) and the rumble filter before being sent to the closed loop amplifier. The signal is compared with the accelerometer signal and the difference amplified via the servo amplifier and subwoofer power amplifier to drive the speaker. Within the servo amplifiers are compensation networks to provide stable loop operation. Loop gain is adjustable as well as output balance between the satellites and subwoofer. Common op amps are used for the satellite crossover and servo controls as shown in the circuit diagram (**Fig. 1**).

SATELLITES

Jac: The satellites are two-way, closed-box towers using an old pair of Dynaudio D-28 tweeters and Dynaudio 17W-75 woofers. The horn-loaded tweeters with their dome recessed into the speaker align the bandpass acoustic centers on the same baffle when measured at 1m. The satellites are crossed with a 4th-order acoustic, passive crossover. The enclosures

are stuffed with fiberfill with a passive rolloff at 75Hz. **Photo 1** shows a satellite speaker with the subwoofer speaker.

SUBWOOFER

Jac: A servo subwoofer needs to be a closed-box design. The box size is established to provide a rolloff at low frequencies that will complement the servo design. This calls for a 6 to 10dB down

point slightly below the lowest frequency for which you are designing. In general, you want the driver and enclosure to combine with a Q_{ts} lower than 0.7.

With a speaker that is overdamped, the phase change at the lower frequencies is smoother and the phase angle limit for the servo loop occurs at a lower frequency. More discussion of this later, but it is important to select a driver and enclosure

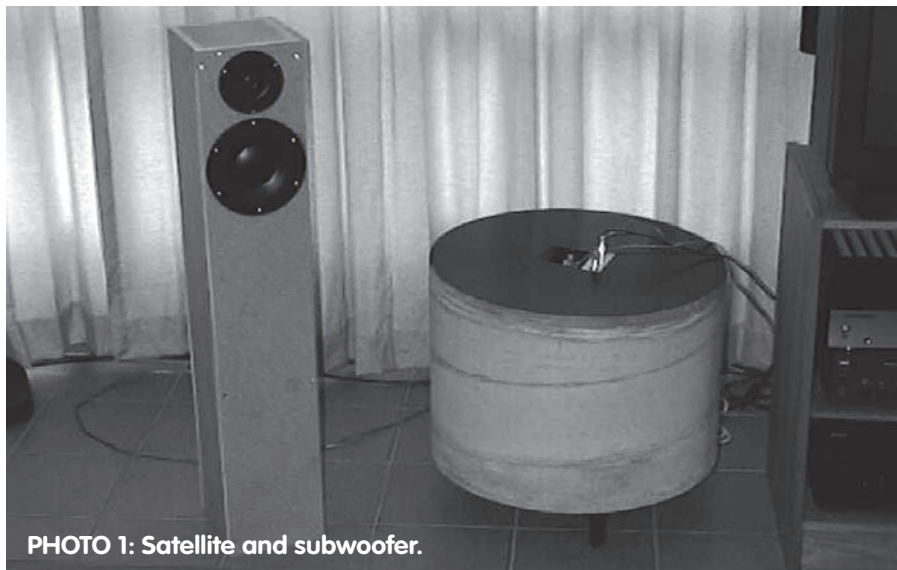
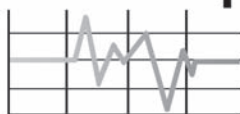


PHOTO 1: Satellite and subwoofer.

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design that is well damped.

In choosing the driver for a servo subwoofer, you also need to make sure you have enough power rating plus enough cone area and travel to reach the loudness at the lowest frequency you are targeting. In my case, I targeted 95dB at 20Hz and 1m. I chose the NHT 1259 for its long travel and power handling. I started with the enclosure with a forward-facing driver for appearance reasons, but as many have found out, you can't mount the NHT 1259 horizontally without the voice coil rubbing. After building the speaker enclosure, I changed the orientation to down firing by adding 4" legs to the speaker baffle (**Photo 2**).

ACCELEROMETER

Art: When I was building my servo subwoofer, it wasn't easy to get a lightweight accelerometer that I could attach to the voice coil of the driver. After some playing around, I decided that I could use the piezoelectric disc from a tweeter as an accelerometer. The piezo disc was mounted on a small post that was attached to the dust cap. The mass of the disc would cause it to flex under the motions of the speaker.



PHOTO 2: Subwoofer details.

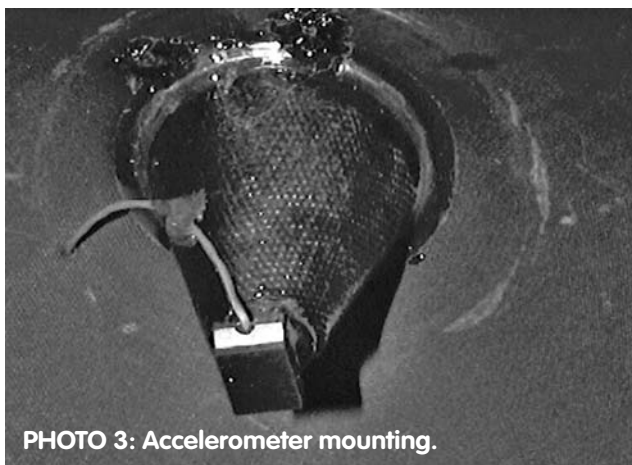


PHOTO 3: Accelerometer mounting.

When the disc flexed, it generated a small voltage that I could amplify. That was my accelerometer. My son had it much easier.

Jac: By the time I started my project, I could find a small accelerometer in the Digi-Key catalog (ACH-01-04) for less than \$25. It had a higher resonant frequency than my Dad's homemade version and had built-in buffering electronics.

I made my accelerometer mount from a cone-shaped, paper drinking cup, fiberglass, and some balsa wood. The drinking cup was the form and it was cut to the size of the voice coil on one end and the size of the accelerometer on the other end. The balsa wood was shaped to fit in the accelerometer end to provide a flat mounting base. The whole assembly was stiffened by glass cloth and 5-minute epoxy that also held the mount to the voice coil and the accelerometer to the mount. The mount became the new dust cap (**Photo 3**).

Art: The accelerometer and its mounting need to have a relatively high natural frequency compared to the frequency of your low-pass crossover. If the frequency isn't high enough, the servo loop will have a phase shift of 180° and at a high enough gain level that the amplifier will tear the speaker apart.

In Jac's case, the accelerometer has resonant frequency of 35kHz and he figures his mounting system resonance is around 3kHz. This provides sufficient margin

while designing the servo. If the resonance is 1kHz or lower, you may need a filter to remove the accelerometer resonance from the loop response.

SUBWOOFER ENCLOSURE

Jac: I had been thinking about non-traditional enclosure designs and had come to the conclusion that both shape and material have important roles in enclosure design. I decided to make this subwoofer enclosure as a cylindrical section because the cylinder is naturally stiff. I made the cylinder section from fiber-reinforced concrete, which has very high stiffness and is quite well damped when compared to typical wood products. **Photo 1** shows the finished enclosure.

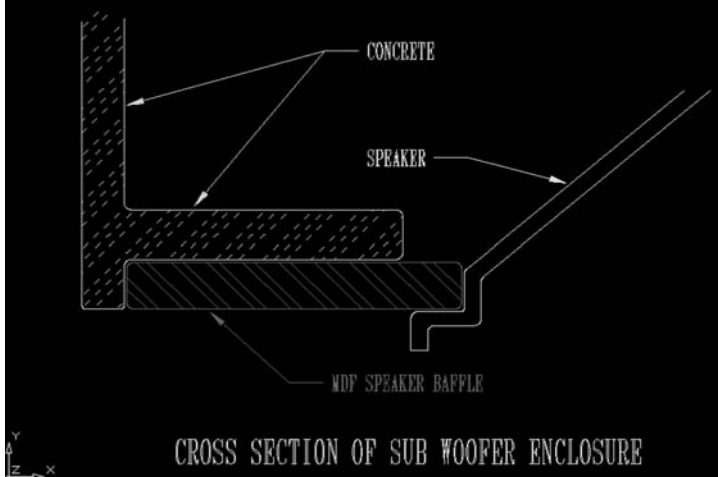
I made a mold using the thick cardboard tubes (Sono Tubes) that are used to cast concrete lamp bases in parking lots. MDF (medium density fiberboard) circles of different diameters made up the bottom of the mold, held the inner and outer tubes concentrically, and formed a rebate for the speaker baffle to fit flush with the concrete cylinder (**Fig. 2**). I used 20" and 22" ID cardboard tubes, 14½" long to give an internal volume of about 2.5ft³. Because the cardboard tubes are ⅜" thick, the 20" ID tube has an OD of 20¾". The result is a concrete wall thickness of ⅝" in the concrete cylinder that has worked very well for sound radiation.

In addition to the concrete cylinder, a concrete baffle is cast near the bottom end of the cylinder. This concrete baffle supports the MDF baffle where the speaker attaches and makes the end of the cylinder nice and stiff (**Fig. 2**).

The top of the cylinder is made of

FIGURE 2: Subwoofer detail construction.

SB-2633-2



CROSS SECTION OF SUB WOOFER ENCLOSURE

MDF 1½" thick. The enclosure is stuffed with fiberfill and threaded rods go all the way through from the speaker on the bottom to the top. Without the driver, the enclosure weighs 65 pounds, which is not too bad for concrete.

SERVO DESIGN

Art: A servo system uses the output speaker accelerometer as a measure of how well the amplifier is driving the speaker. Refer to **Fig. 1**. While testing the servo loop, use the normal input to the loop summing amplifier (v11) and look at the accelerometer signal as the output (disconnected at A, the input to the loop summing amplifier). A sweep of the range of frequencies will show how well the speaker can reproduce the input.

Figure 3, the open loop design curves, shows this response as amplitude and phase shift between the input and the output. The curves show an amplitude bandpass and phase shift *after* we had installed the loop high-frequency compensation filters (amp v13) that attenuate the high frequencies. An important part peculiar to this curve is the speaker/accelerometer response around 660Hz. As discussed below we provided a low-pass filter that would attenuate the loop response to cause this "noise" at 660Hz to be near -10dB gain in this open loop curve. This gave us a desirable margin in the loop design.

The other part of this response curve, **Fig. 3**, is the phase shift that occurs as the frequency is changed. At 80Hz, the phase shift is 180° *out of phase*. This means that the output could cancel most of the input when added together in the summing amplifier (v11). This is desirable because the servo will, in closed loop, allow just enough difference between the input and output signals to drive the speaker at the level we want for the input level. This difference between the output and input depends upon the loop gain—the higher the loop gain, the smaller the difference.

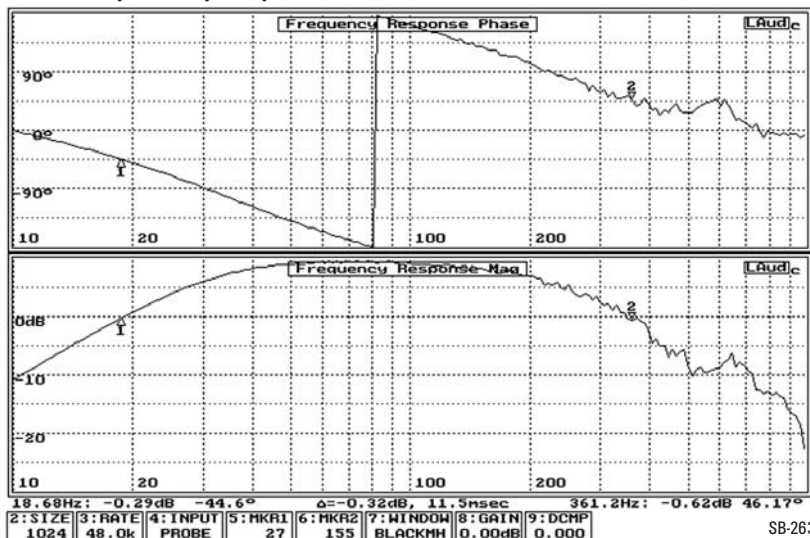
The next part of this design requires you to look at the phase shift between the two signals at the high and low frequencies. If you allow the two signals to sum when they are in phase and the gain is greater than 1 (>0dB), you will allow the signals to reinforce themselves causing an unstable (or blow up) condition. Adjusting the loop gain does not change

the loop phase characteristics. We chose to adjust the loop gain so that the 0dB gain point is about 135° phase shift from the central phase (180°). The curve shows this to be 45° at both the high and low frequencies.

The open loop curve may be a different shape, due to a different tuning of the speaker in the enclosure, or at the high frequency end with different high

frequency compensation. In either case as you adjust the loop gain, you should select the end of the curve that reaches the 135 (45°) first, and accepting the lower phase shift at the other end (90° for example) so as to not exceed 135° phase shift anywhere in the frequency range. The resultant closed loop response will be stable, but the shape of the closed loop curve will be different. The end with the

FIGURE 3: Open loop response curves.



SB-2633-3

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lower phase shift will be more damped and the performance of the speaker system will be changed accordingly. The new performance of the speaker may be exactly what you want. It is not wrong but just a decision on the part of the system designer.

CLOSING THE LOOP

Figure 3 shows the open loop gain and phase curves of the servo prior to closing the loop or physically connecting the accelerometer signal to the summing amplifier at A. **Figure 4** shows the servo

loop after closing it. The loop includes the summing amp, the loop gain amp, the high frequency compensation filter, the power amplifier, the speaker, the accelerometer, and amplifier—all affecting the response. I measured the closed loop gain as output/input (accelerometer voltage as representing the speaker output/input voltage to the summing amplifier). A speaker acoustical output (microphone) could have been used for the output signal.

1. Note that in **Fig. 4** we have peaks in

response at 18Hz and 400Hz. These result from the phase shift and gain that occur at these frequencies, where you see some regeneration, but not enough to be unstable. We decided this was acceptable.

2. At about 660Hz there is a peak that could cause trouble. It also appeared in the open loop curves, so you can assume that it's a cone resonance of some kind. We decided to make sure that the crossover lowers the peak to at least 20dB below bandpass. The 4th-order crossover

FIGURE 4: Closed loop response without filters.

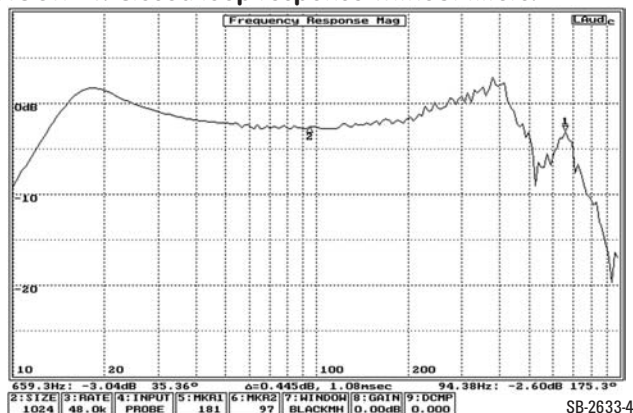
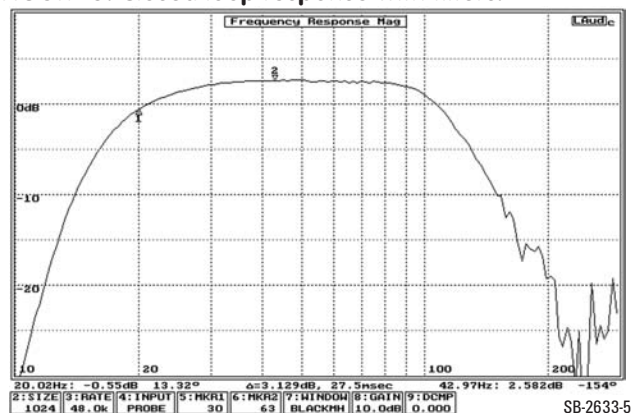


FIGURE 5: Closed loop response with filters.



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By Jack Darr

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at 105Hz in the input amplifier (v7 and v8) was successful in lowering the peak to 20dB down. **Figure 5** shows the closed loop response with the rumble and crossover filters.

3. There was a rising output as the frequency drops from 80Hz to about 18Hz (**Fig. 4**). In this case, the rumble filter in the input amplifier is used to shape the low-end response.

Figure 5 shows the full subwoofer response including the crossover and rumble filters. As you can see, the result is very smooth with an F3 of 20Hz and a crossover frequency of 105Hz.

SUBWOOFER AMPLIFIER

Jac: In addition to the servo circuitry, a high power amplifier is needed to drive the subwoofer. How much power you need will vary depending on your driver, enclosure, and low frequency target. The amp must provide power for the loudness you want at your low frequency and to make up the difference in loudness between the passive response of your speaker and your desired response curve.

For example, suppose your driver has a sensitivity of 87dB at 1W/1m and you want a maximum loudness of 96dB at 1m and 30Hz. To get 96dB from the passive speaker at 100Hz, you need to add 9dB above the 87dB. Because every 3dB doubles the input power, you will need 8W of input power available to get to 96dB at 100Hz.

If the passive speaker is 12dB below bandpass at 30Hz, the servo system needs to provide power to make up that difference. From 8W at 100Hz, you need 128W at 30Hz. Because you need some transient headroom and amplifier damping, you should probably spec the subwoofer amp at a minimum of twice the required power.

I chose to use an Opti-MOS kit amplifier from G. Randy Slone that is rated at a conservative 500W. Mr. Slone is no longer selling these kits, but he has written several excellent books (McGraw-Hill) on high-power amplifiers that include circuit artwork and information to build excellent amps including the Opti-MOS. My power amplifier is shown in **Photo 4**, with the servo amplifier boards on top, the power amp board below, and the power supply.

CONCLUSION

Jac: Building a servo subwoofer system is certainly no easy project. We hope this article gives you an idea of some of the challenges and trade-offs in designing and building a servo subwoofer.

How does it sound? It's the deepest and cleanest bass I've ever heard. The

satellites even sound better—I suppose because they aren't trying to produce frequencies below 105Hz. There is no way for me to compare it to other possibilities. All I can say is that all of my friends who have heard it are impressed, and I'm spending much more time listening to music. **aX**

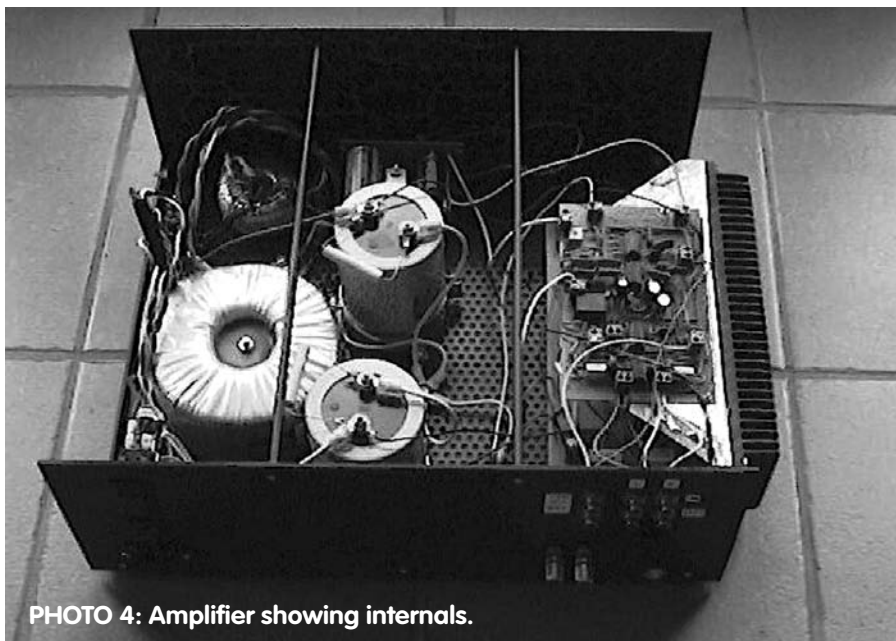


PHOTO 4: Amplifier showing internals.

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ACOustics Begins With ACO™

A Balanced Nonoversampling Retro DAC

This high-end DAC design is a perfect blend of both contemporary digital technology and old vacuum tube and transformer techniques.

By Yury Gelgor

Why should audiophiles pay attention to digital analog converters (DACs)? Well, in my case, when cheap hi-fi VCRs sound better than three times more expensive CD/DVD/SACD players, it is a good time to take note. And this was after I had already made a pretty good amplifier with speakers and could still hear a difference.

This article shows how to construct a DAC that is an upgrade of a previous one I have posted on my website www.geocities.com/yury_g. However, this article is not just about the upgrade; it is about the whole RetroDAC (**Photo 1**). This DAC is not a cheap upgrade of a CD player, but a pretty expensive unit that stays below my reasonable limit just because I did not use silver-wired transformers, exotic capacitors, and other fancy components—with one exception.

WHY BALANCED?

The 1950s–1970s were an era of audio reproduction when a very high standard was set in the studio and nobody knew about good audiophile interconnect cables. Simple twisted wires could deliver that quality. How so?

The answer is simple: 600Ω balanced transformer outputs. This schema has many advantages such as ground loop breaking, reduced hum, and natural low output impedance.

The first disadvantage of such a solution is the demand for a high-quality output transformer, which comes with a high price, of course. Some famous microchip producers make ICs of “transformer-like” balanced line transmitters/receivers; however, the quality of transformer schema remains beyond compare.

Another problem with the trans-



PHOTO 1: The completed DAC.

former is DC offset, which is simple to avoid by using a push-pull amplifier that reduces even distortions. You need a phase splitter to feed push-pull schema. I think the best way to build this phase splitter is to use additional DAC chips in each channel to produce two signals with 180° phase shift. I will explain this presently.

WHY NONOVERSAMPLING?

Let me remind you first what sampling is. Studio recording devices measure the voltage of audio signals 44,100 times per second and record those numbers on CD. On the listening side, a CD player reads recorded numbers and sends them to the DAC for conversion into voltage. Those numbers that appear 44,100 times per second are samples, and only they represent true values of the signal in some moments of time. No true information is available between the moments when the samples were taken.

Oversampling is an attempt to calculate those signal values using a sine approach to shape. It is probably good to measure single signal distortions; however, it is not correct at all. Nobody can calculate music, which is obviously not a sine signal.

Another reason for not using oversampling: the higher the oversampling, the higher the jitter level. It means that two times oversampling DAC has two times higher jitter than a nonoversampling one with the same CD/DVD mechanism (transport).

THE DIGITAL PART: KEEP IT SIMPLE!

A transport processor reads audio information from CD/DVD and sends it as a bit stream to digital output. The protocol (the way of bit stream coding) is called S/PDIF (Sony/Philips Digital Interface Format). Digital output of the transport could be organized in one of two ways: coaxial or optical. Optical is called Toslink (Toshiba optical link). I prefer coaxial connections between the transport and DAC.

As a true hi-end lover, I am sure that the simpler the schema the better. I also think that the best resistor is no resistor at all, the best capacitor is no capacitor, and so on. So you know that when I use a part in this DAC it is really necessary and essential.

Therefore, the first task of the digital part is to receive this S/PDIF signal from the transport, synchronize to it,

recognize the audio format, and convert it to signals the DAC chip can “understand.” IC1 takes care of all that. It recognizes regular CD 44.1kHz format, practically all other professional formats, and most DVD-audio formats.

Figure 1 shows the simplified digital part of the DAC. IC1 produces three important signals:

- SCK-clock signal—For a regular CD it has a frequency of $44.1\text{kHz} * 64 = 2822.4\text{kHz}$

- SDATA—Data for recording to input register of the DAC and the proportional output signal.

- FSYNK—High/low level here shows that data on SDATA output is for the left/right channel, respectively.

IC3 and IC4 are precision twin DAC chips. Each has the following inputs: D+, D- (data inputs for no inverted and inverted shoulders), L+, L- (latches for recording of data into register), and CLK (input clock).

It takes one inverter IC2B to separate information for the right channel. It also takes another inverter IC2A to get inverted data for both. This is it, no more parts. The very smart receiver IC1 makes everything that’s needed to bring necessary signals to the set of precision DACs IC3 and IC4 and only two inverters to separate the bit stream for the left and right channels and organize completely balanced schema.

DIGITAL PART 2: UNDER FIRE

Since I put the previous version of DAC on my website, I have received several e-mails from folks who criticized the digital part of my schema. Two subjects under fire were phase shift and code for inversed shoulder:

1. Phase Shift. At the recording, both channels’ samples are made at the same time. CS8414 has serial outputs: the first goes out the left channel, then the right channel with 32 clock-beatings delay (signal FSYNK), which is $T=1/(2 * F_{\text{smpl}}) = 1/(2 * 44100) = 11.34 \text{ microseconds}$, where F_{smpl} is the sample frequency.

It produces the phase shift between the channels = $360 * F * T: 360 * 1000 * 0.00001134 = 4.08^\circ$ at 1kHz and $360 * 20000 * 0.00001134 = 81.7^\circ$ at 20kHz.

It is definitely not significant at 1kHz

and hard to say whether it is possible to hear such phase shift at 20kHz. Maybe somewhere in the middle? When using headphones? I do not think so, but I believe that 32-bit shift register, which needed to correct this problem, can bring more digital noise to the power supply that caused more jitter. So I think the effect of using such a register will be negative.

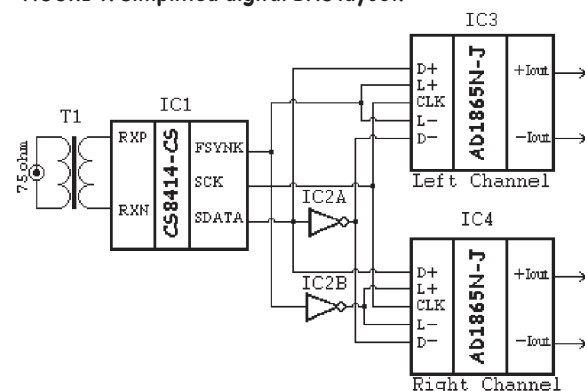
2. Inversed code versus complementary code for inversed shoulder.

Forming data for an inversed shoulder DAC strictly with the inverter is not correct: to get a correct 16-bit complementary code you must add one after the inversion of code; otherwise, you always have one bit misbalanced. It is the same linear signal as in “positive” shoulder, but 180° inversed and one

bit shifted to the negative side. For example, when no signal is applied, the code for “positive shoulder” is 0000,0000,0000,0000 and for “negative shoulder” is 1111,1111,1111,1111. This is exactly one bit misbalanced: just add one to the negative shoulder code and you get “all zeroes.” There is any signal or no signal at all.

The maximum output current for the DAC AD1865 is 1mA, which obviously happens when the signal is maximal. So this misbalance is $1\text{mA}/8192 \text{ bit} =$

FIGURE 1: Simplified digital DAC layout.



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0.00012mA. About a dozen chips schema is needed to correct this problem, and, of course, the twin triode has much bigger misbalance, so it's best to forget about this "one bit problem." For me, the dozen additional IC is the same reason to be afraid of jitter.

ANALOG PART: EVEN SIMPLER!

Figure 2 shows one channel schema of my DAC. The twin DAC is "pumping" the balanced signal into R3, called "I/V resistor," and T2 ("I/V transformer") primary where I/V conversion is made. T2 has a ratio of 1:6.4 + 6.4. That signal goes to the twin triode balanced schema, which is also "pumping" the primary of T4. The secondary of T3 is the 600 Ω output of the DAC. This schema is very simple, but some parts of it deserve further comment.

First, consider the value of the I/V resistor. There are so many different opinions about it! Some folks reduce this value to 10 Ω , others increase it to 1k. So what? Some say that Burr-Brown DACs work better with smaller values and 25–40 is fine. But any attempt to use values over 100 Ω causes sound similar to signal clipping. Some also say Analog Devices

DACs are more tolerant to high values of this resistor, which could be up to 600 Ω .

So I changed the value of this resistor to listen to the difference. Obviously, the higher its value, the bigger the output signal. And to my ear, values less than 100 Ω sound worse: drums, explosions, all high energy short sounds play better within 200–400 Ω rather than 25–50. The low value of that resistor suppresses dynamics of sound.

However, the higher the value of this resistor, the higher the inductance of the T2 primary must be: for 25 Ω it is fine at 0.5H, for 250 Ω it is 5H. So the primary inductance of the transformer sets the limit of this resistor.

Now consider the high-frequency response of transformer T2. Every transformer has high-frequency (HF) resonance (**Fig. 3**, F2). The higher the frequency of this resonance, the better this transformer is for normal amplifiers, but things are a little bit different for the DAC.

Figure 4 shows a signal waveform at the I/V resistor. Instead of a smooth wave of signal, one step follows another. This is the way it works: the DAC chip converts incoming code into analog and keeps its value until the next one. Each code (sample) follows another with a frequency of 44.1kHz, and the output signal includes a large amount of this frequency. And yes, your guess is right: it harms the signal,

and the I/V transformer must have HF resonance beyond that.

ANALOG PART 2: RLC FILTER VS. TRANSFORMER

An important role of the transformer is to be a filter for reducing the sample frequency level. A regular RLC filter just changes impedance and reflects incoming energy back to the source (DAC output). In turn, the source again reflects the rest of the signal to the filter. The signal becomes reflected many times, losing quality and sharpness exactly like a TV picture without a properly adjusted antenna.

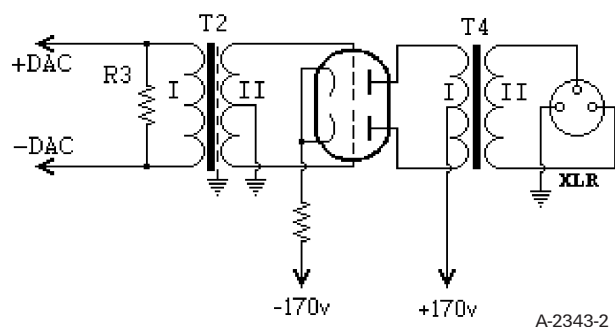
Look again at **Fig. 3**. A correctly made transformer has a flat response at 20kHz (F1), HF resonance (F2) about 28–33kHz, and 44.1kHz is supposed to be on high-frequency rolloff (F3). A transformer such as this mostly turns that energy into heat and consumes it, which makes the signal's shape much clearer (**Fig. 5**).

POWER SUPPLY

Design of the power supply (**Fig. 6**) is as important as signal audio or the digital part. After many experiments, I think that if I need several voltages, it is better to use a separate transformer for each one, not just separate secondary windings. Furthermore, the best transformer is no transformer at all when I need about 150 DC voltage. That is why I use +170V and –170V one-wave rectifiers for tube output. Note: This is dangerous, so you should take some precautions to avoid electrocution.

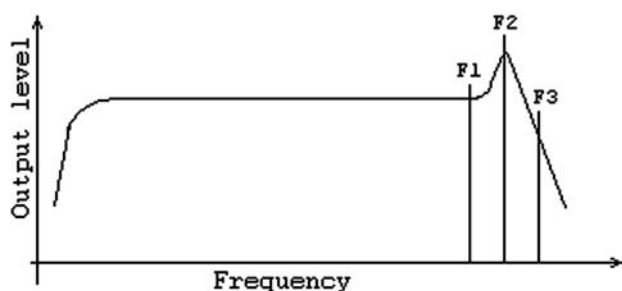
If you have no high voltage experience or are not sure about the result, just use a 1:1 ratio transformer for tube plates. Under no circumstances should you use

FIGURE 2: Analog portion of DAC.



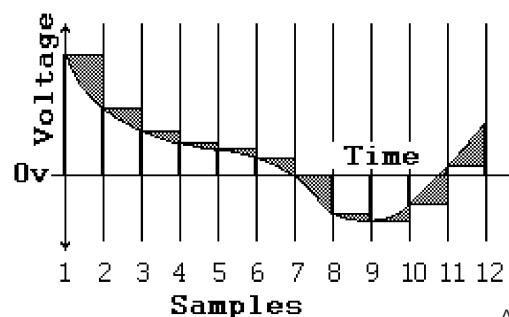
A-2343-2

FIGURE 3: Typical frequency response of transformer.



A-2343-3

FIGURE 4: Waveform and samples.



A-2343-4

A schema with “Alert” neon lamp shows when polarity is wrong, and “Polarity” switch allows you to change polarity. It is a good idea to use a special AC socket for this kind of schema. Working with such schema is simple:

- The other sources of the power supply are pretty simple. Using ferrite shield beads for tube heaters is something that you may not see too often. They are used to reduce the chance of high-frequency parasitic oscillation.

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on the bottom side of this frame to make it solid. So, mechanical design is also very simple!

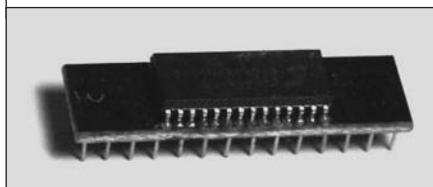
You probably already found the BNC connector near the S/PDIF RCA input. It is not connected and is for future use only.

It is very important to make a good ground. See **Photo 4**. Output ground (OuG) is made of 8 gauge (about 3.5mm) solid copper wire. A ground star (where all grounds come together and connect to the metal chassis) is shown in the white circle.

TUNE-UP PROCEDURE

1. Remove IC1-IC4 from the sockets. Do not connect any interconnect cables. Remove all tubes.
2. Plug in. Make sure the "Alert" light is off. Flip "Polarity" switch to another position when "Alert" light is on.
3. Use potentiometer R119 to set output voltage of tube heater regulator to 6.3V on pins 4 and 5 of tube

PHOTO 3: IC1 on SOIC-to-DIP adapter.



sockets Vt1 and Vt2.

4. Check out voltages on capacitors C111, C113, and C116. They must be +5V, +5V, and -5V, respectively.
5. Check out all necessary voltages and grounds on each chip according to schema. (For example, pins 8, 17, 21, and 23 must be 0V and 7, 16, 18, 22, and 24 must be +5V in socket of IC1. The chip is not inserted yet!)
6. Unplug unit, put in Vt101. Plug in. Make sure you have about -170V on pins 3 and 8 and +170V on pins 1 and 6 of tube sockets Vt1, Vt2.
7. Unplug unit, put in tubes Vt1 and Vt2. Plug in. Make sure you have about +5.4V on pins 3 and 8 and +93V on pins 1 and 6 of tube Vt1, Vt2.
8. Unplug Unit. Insert IC1-IC4. Do not forget to use Electro Static Discharge (ESD) bracelet.

PHOTO 4: Output Ground (OuG).

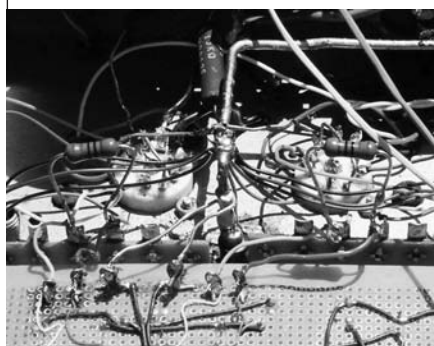
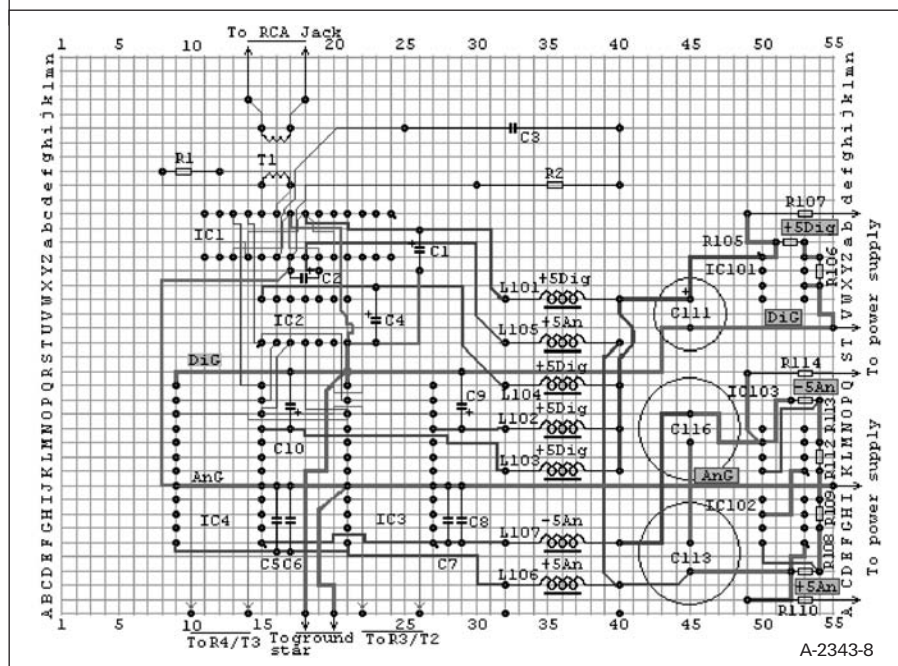


FIGURE 8: Digital board layout.



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on the bottom side of this frame to make it solid. So, mechanical design is also very simple!

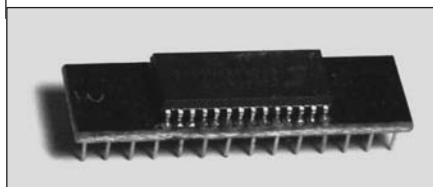
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TUNE-UP PROCEDURE

1. Remove IC1-IC4 from the sockets. Do not connect any interconnect cables. Remove all tubes.
2. Plug in. Make sure the "Alert" light is off. Flip "Polarity" switch to another position when "Alert" light is on.
3. Use potentiometer R119 to set output voltage of tube heater regulator to 6.3V on pins 4 and 5 of tube

PHOTO 3: IC1 on SOIC-to-DIP adapter.



sockets Vt1 and Vt2.

4. Check out voltages on capacitors C111, C113, and C116. They must be +5V, +5V, and -5V, respectively.
5. Check out all necessary voltages and grounds on each chip according to schema. (For example, pins 8, 17, 21, and 23 must be 0V and 7, 16, 18, 22, and 24 must be +5V in socket of IC1. The chip is not inserted yet!)
6. Unplug unit, put in Vt101. Plug in. Make sure you have about -170V on pins 3 and 8 and +170V on pins 1 and 6 of tube sockets Vt1, Vt2.
7. Unplug unit, put in tubes Vt1 and Vt2. Plug in. Make sure you have about +5.4V on pins 3 and 8 and +93V on pins 1 and 6 of tube Vt1, Vt2.
8. Unplug Unit. Insert IC1-IC4. Do not forget to use Electro Static Discharge (ESD) bracelet.

PHOTO 4: Output Ground (OuG).

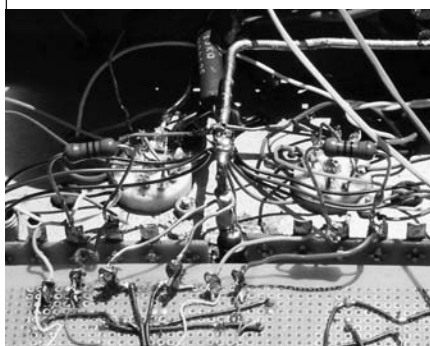
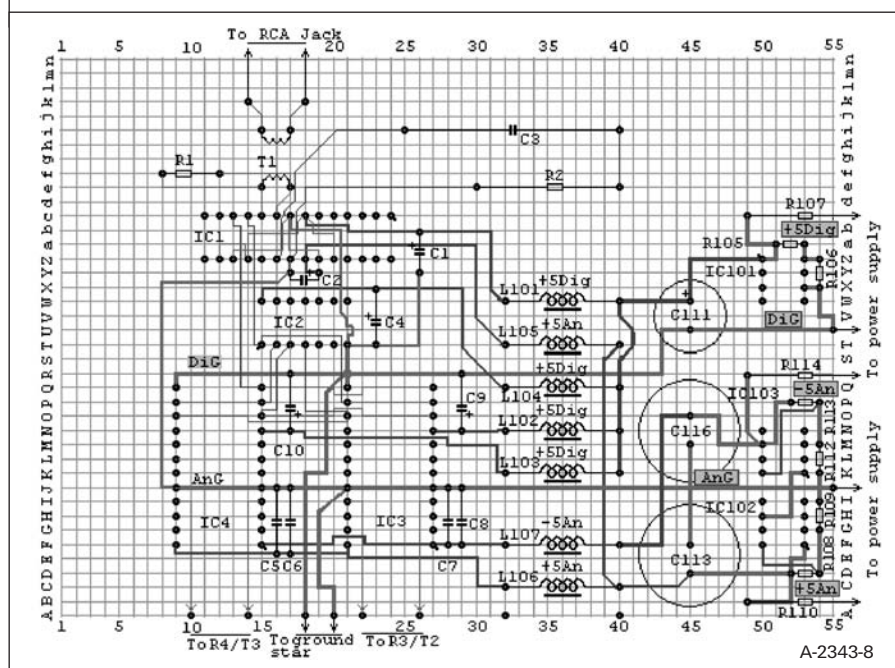


FIGURE 8: Digital board layout.



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9. Plug in interconnect cables, plug unit in.
10. Enjoy! Do not forget to program your CD/DVD player according to its user manual. Many of them have S/PDIF coax output initially turned off.

UPGRADE

I did not change the schema of the DAC during the upgrade with one exception: I added Black Gate capacitors C102 and C107. All other changes are:

1. Recommended for CS8414 Scott SPDF transformer, I replaced with Audio Note TRANS-280. This

is the only transformer with silver wiring.

2. I replaced Sowter 8347 T2 and T3 transformers with custom-made Sowter 9762.
3. I replaced resistors R1, R2, R108, R109, R112, R113 with Audio Note Tantalum.

TABLE 1: Parts List

COMPONENT	VALUE	MANUFACTURER	SUPPLIER
BR101	5A	Any diode bridge 5A or more	
C1, C2, C4, C9, C10	100µF 10V	Black Gate PK	Parts ConneXion
C3	68nF 100V	Multicap RTX	Parts ConneXion
C5, C7	22µF 6V3	Black Gate NX	Parts ConneXion
C6, C8,	4n7 500V	Silver Mica	www.newark.com
C101, C106			
C102, C107	68µF 350V	Black Gate NH	Parts ConneXion
C103-C105, C108-C110	150µF 350V	Black Gate VK	Parts ConneXion
C111	220µF 16V	Black Gate FK	Parts ConneXion
C112, C114, C115, C117, C118	1000µF 25V	Black Gate Std	Parts ConneXion
C113, C116	1000µF 25V	Black Gate NX	Parts ConneXion
C119	100µF 16V	Any	
C120	1µF 16V	Any	
C121	23000µF 16V	Mallory	www.newark.com
D101	NTE174	NTE (see note)	www.mouser.com
D102, D103	HFA08PB60	International Rectifier	www.newark.com
D104, D105, D106, D107	NTE174	NTE	www.mouser.com
F101, F104, F105, F106	0A25 250V	Any slow fuse	
F102	1A 250V	Any slow fuse	
F103, F107	0A5 250V	Any slow fuse	
L101-L107	47µH	Any RF coil	
LA101	Red Neon Assy	272-712	
LA102	Green Neon Assy	272-708	
R1	110Ω 0.5W	Audio Note Tantalum	Parts ConneXion
R2	470Ω 0.5W	Audio Note Tantalum	Parts ConneXion
R3, R4	220Ω 1W	Audio Note Tantalum	Parts ConneXion
R5, R6	22k 12W	Mills MRA12	Parts ConneXion
R101, R103	2k2 12W	Mills MRA12	Parts ConneXion
R102, R104	1k 12W	Mills MRA12	Parts ConneXion
R105, R106	1k5 0.75W	Caddock MK132	Parts ConneXion
R107	100Ω 12W	Mills MRA12	Parts ConneXion
R108, R109, R112, R113	1.5k 0.5W	Audio Note Tantalum	Parts ConneXion
R110, R111, R114, R115	75Ω 12W	Mills MRA12	Parts ConneXion
R116, R117	220Ω 2W	Any	
R118	120Ω 0.5W	Any	
R119	500Ω 1W	Any potentiometer	
IC1	CS8414-CS	Cirrus Logic	www.newark.com
IC2	74HC04	Any	www.newark.com
IC3, IC4	AD1865N-J	Analog Devices	www.newark.com
IC101-IC103	TL431	Any	
IC104	LM338T	Any	
SB101-SB103	250Ω at 100MHz	Fair-Rite	www.newark.com
T1	TRANS-280	Audio Note S/PDIF	Parts ConneXion
T2, T3	Sowter 9762	Sowter	www.sowter.co.uk
T4, T5	Sowter 8650	Sowter	www.sowter.co.uk
T101	185C10	Hammond	www.newark.com
T102, T103	185C24	Hammond	www.newark.com
T104	185C16	Hammond	www.newark.com
Vt1, Vt2	6H30Pi	Russia	www.tubestore.com
Vt101	5Y3		www.tubestore.com

RetroDAC Versus

Audio Note DAC 5.0

The main differences between RetroDAC and Audio Note 5.0 are:

1. I used two DACs to produce differential signal with no grounded primary tap. With this tap, schema is not completely differential, sounds worse (to my ear), and looks more like two Audio Note DACs, working in push-pull mode.
2. I did not use filters, described in the Audio Note patent (US # 5,420,585). Balanced schema allows you to diminish filter to the same I/V transformer, which is obviously better in my opinion.

So which DAC sounds better? I do not know. I never listened to DAC 5.0. It is not a competition, but just another parallel universe.

HOW DOES IT SOUND?

If you expect to discover powerful bass, which is going to blow you away, or super highs—this DAC is not for you. However, if you enjoy big classic orchestra or chorus, “smart rock” such as Pink Floyd or Genesis, or big jazz bands—in short, any music performed with good musicians and many instruments—you will be surprised. RetroDAC easily reproduces the most complex music. I hear many new sounds, instruments, and voices in records I know perfectly well.

RetroDAC also has an outstanding dynamic of sound. It reproduces well loud and faint sounds at the same time

ACKNOWLEDGMENT

I want to express my appreciation to Mr. Brian Sowter, who was very cooperative during about a month-long exchange of opinions over e-mails, regarding construction and parameters of a custom Sowter 9762 I/U transformer (Sowter Audio Transformers, England, www.sowter.co.uk). The result of those conversations exceeded my expectations.

with minimum interruptions. The same feature gave another interesting result: a depth of sound picture. In many cases, especially in movies, you may judge how far the sound is from you.

Even my wife admitted that now she can listen to the music for a long time

without tiring. RetroDAC brings the mood of music or movies. Horror or joy, happiness or sadness—no matter what.

And do not forget, please! You must use a really good amplifier and almost perfect speakers to get full advantage from RetroDAC. Those two compo-

nents of your audio system are most important. Sound cannot be significantly improved just by using any DAC if you use average hi-fi. **ax**

Yury Gelgor was born in the Ukraine and received his bachelor's degree in industrial engineering. He is an experienced industrial electronic technician, electronic engineer, and computer programmer. He moved to the U.S. in 1994. Both a hi-fi and tube-sound lover, his dream is to find a job that matches his hobby.

LAST MINUTE UPGRADE

It has now been some time since I've written "A Balanced Nonoversampling Retro DAC," and I could not stay off my DAC and just listen, so I made another upgrade I want to share with you.

First, I completely removed negative power supply for tube bias. It means the following parts were completely removed: F104, D101, C106, C107, C108, C109, C110, R103, and R104. Resistors R5 and R6 are now connected not to 115V power supply, but directly to Output Ground (OuG). New values of R5 and R6 are 2.2k 1W and they are tantalum Audio Note resistors now.

Second, I replaced capacitors C1, C2, C4, C9, C10 with Black Gates NX 100 μ F 6.3V and put jumpers instead of coils L101–L107.

Third, C102 is removed, C103 and C104 are Black Gate 220 μ F 160V and C105 is 68 μ F 350V now.

All that not only made the schema more simple, but significantly improved clarity and musicality of the DAC. ■

PHOTO 5: Top view. Digital board cover removed.

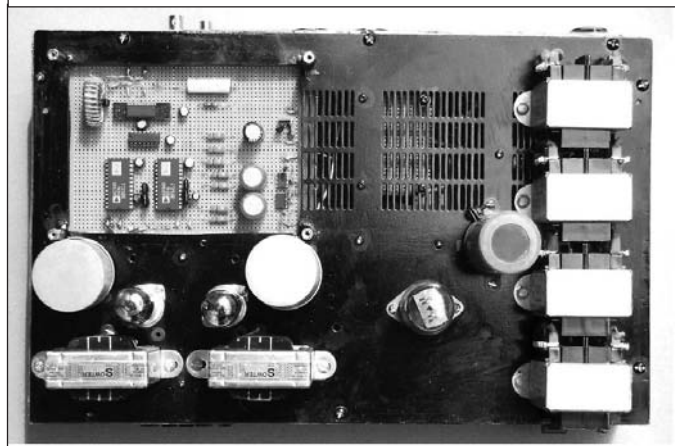


PHOTO 6: Bottom view.

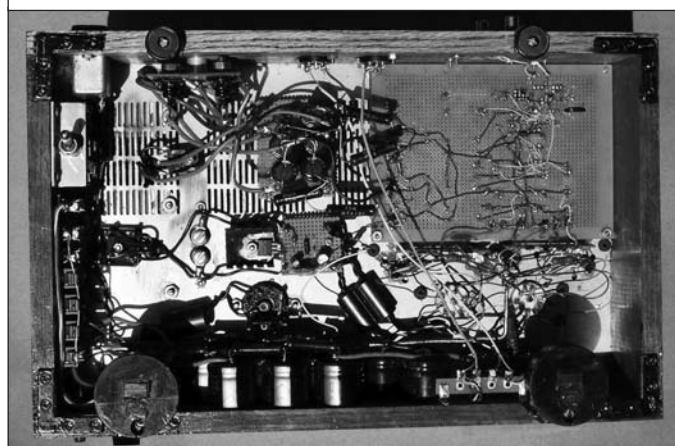


PHOTO 7: Rear view.



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L _e	0.39	S _d	57	SPL	85.9



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Renovating the Scott 222C

Part 2: Modifications

By Charles Hansen

[Back](#)

I removed the tubes and stored them in boxes. If you don't have the original boxes, use plain white tube boxes (which you can obtain from Antique Electronic Supply) and write each tube type and its location in the chassis on the box. Wrap the boxes in bubble wrap. Bag and tag any parts and hardware you remove in the course of work and store them with any other components. Do not throw anything away until you again have a fully functional unit.

A number of websites list recommended changes to the Scott 222 series. The most common suggestions are:

- Replace all the ceramic-tube paper American Ceracaps with modern polypropylene film caps.
- Replace the selenium bias supply rectifier with a silicon rectifier (mandatory!)
- Update the output tube grid bias pot circuits to the 222D model.
- Replace the aluminum can capacitors with modern versions made by CE Manufacturing.
- Convert the output stages to ultralinear output transformers.
- Use modern speaker binding posts and phono jacks.
- Clean and polish the aluminum chassis.

If you would like to renovate or restore a Scott amplifier with 7189 output tubes, I suggest starting with the 222D. The 299A and 299B also used 7189 output tubes. None of the Scott products used ultralinear output transformers.

Scott made optional wood and metal cabinet covers available for its hi-fi equipment. In my view the heat captured by the cover could shorten the life of tubes and temperature sensitive components such as the electrolytic caps. If child safety is not an issue, I recommend running the amplifier without a cover.

DECONSTRUCTION

Because I would be replacing probably 95% of the components while making extensive modifications to the rear panel, I made "as received" drawings and took detailed photos of the component and wiring installation before doing anything else. I was able to find on eBay the construction manual for the LK-48, which is the Scott kit version of the 222. This was quite helpful in taking the 222C apart in the appropriate manner. I also took digital photos of every step in the modification to assist in documenting the changes.

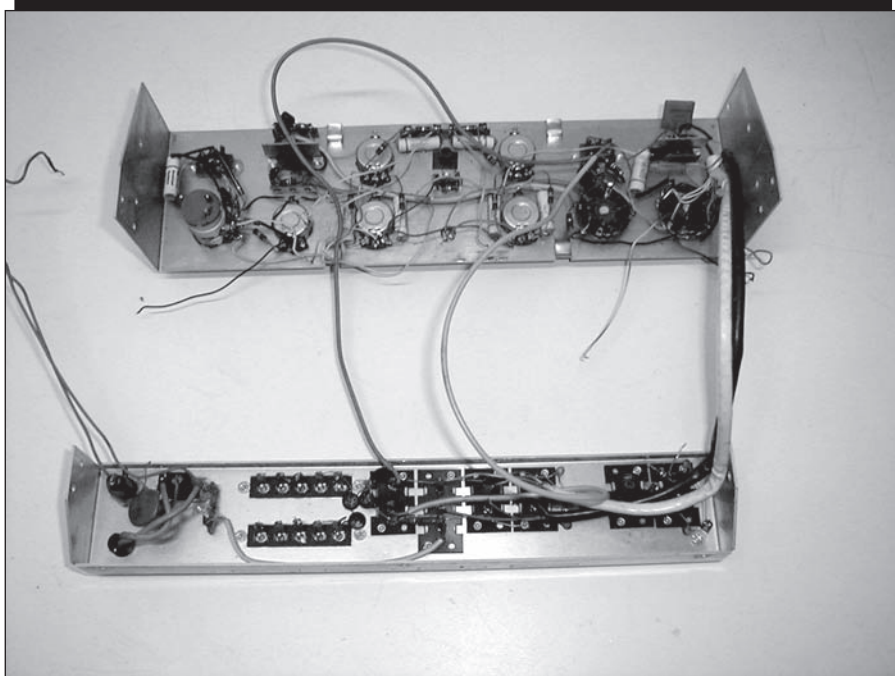
I removed the front and rear sub-chassis sections (**Photo 5**) from the main chassis, and then removed all the chassis-mounted parts that I was going to replace. Make certain you de-solder in a well-ventilated location. The highly reactive solder fluxes used back then, as well as the fine dust accumulated over time, are none too friendly to your lungs.

I use a muffin fan equipped with a charcoal filter for an indoor air purifier.

Switches, sockets, can capacitor insulators, speaker terminals, and jacks are all riveted to the sheet metal, so these rivets will need to be drilled out with a 5/32 drill if you need to replace any of the parts. You can use 6-32 screws, nuts, and lock washers to replace the rivets.

I stripped most of the wiring and components from the main chassis, except the tube sockets, terminal strips, and the three transformers. This was an incredibly tedious job made moreso by the propensity of the dedicated ladies of Maynard who wired the 222Cs to wrap every wire and component lead around the various terminal connections far in excess of the 180° necessary for a good mechanical connection. Most were wrapped at least 360°, and in some instances as much as 720°! And the tail ends were tucked tightly against each terminal so it wasn't easy to unwrap. It

PHOTO 5: Front and rear sub-chassis removed.



meant days of careful struggling with soldering and de-soldering irons, diagonal cutters, and needlenose pliers to prevent breaking off or overheating a terminal on the custom terminal strips that Scott used.

In the end, after breaking a few tube socket lugs, I decided to replace all the 9-pin sockets. Time and temperature had made the copper lugs brittle and beyond saving. I removed the three can capacitors by carefully untwisting the can lugs, and left the insulated mounts in place because they were in good condition. The main chassis—after stripping all the parts but before removing the transformers and doing any cleaning—is shown in **Photo 6**.

I removed the pots and rotary switches from the front sub-chassis for cleaning. The slide switch actuators are color-matched to the gold front panel, so I decided to try to save them by cleaning them with Caig Deoxit. The loudness control is a custom item, so my goal was to save it if possible as well.

The back half consists of a 500k pot tapped at 18k5, with an attached push-pull switch. This section is soldered to the rear of the front tapped 500k pot section and interconnected with a custom shaft to operate the whole stereo assembly. The later Scott 299/LK-72 used a separate slide power switch, making the upgrade to a new dual pot very easy. Note that Noble and Alps do not

PHOTO 6: Main chassis after removing parts and wiring.

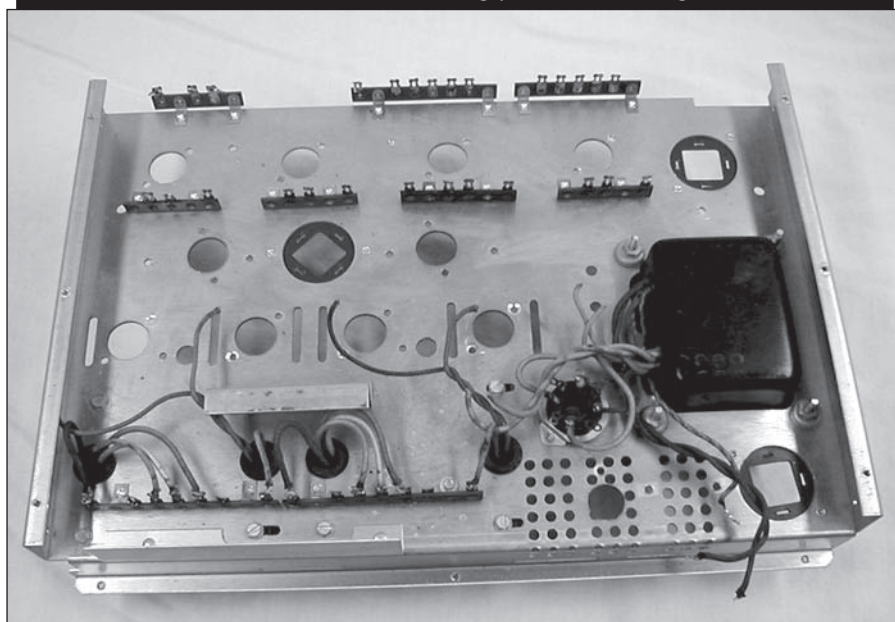
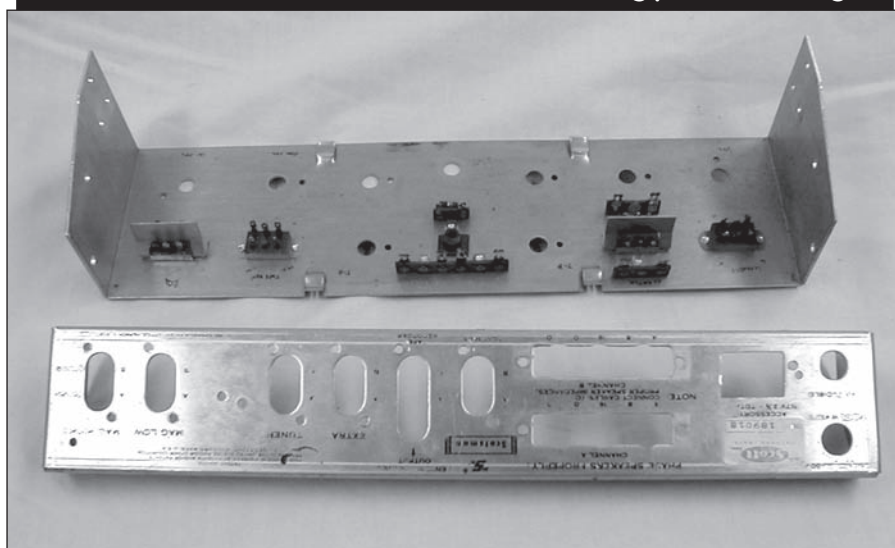


PHOTO 7: Front and rear sub-chassis after removing parts and wiring.



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I removed all the tin-plated phono jacks by carefully bending their tabs. Then I drilled out the rivets, saving the outer phenolic insulators for re-use when I replaced the phono jacks with modern gold-plated jacks. The old jacks were vertically spaced less than $\frac{1}{2}$ " apart, so I'll need to address the problem of clearance for the larger shells of modern

With all the components removed, I measured the capacitance, dielectric absorption (DA), dissipation factor, equivalent series resistance (ESR), and leakage current of the three multi-section electrolytic can filter caps with a Sencore LC102 Auto-Z capacitor-inductor analyzer. Now this may be pure coin-

The two 25 μ F 25V sections (C14, C114) had very high ESR and DA, and the leakage of C114 was very high at 2.5mA. The four 75 μ F 75V sections all had DA readings above 77%, and their leakage current was higher than 35mA. Note that in every instance the capacitance was within specification. The lesson here is that capacitance readings alone will not identify a bad electrolytic capacitor.

Many of the carbon composition resistors were well outside their 10% tolerances (carbon comp resistors change value with temperature, age, and humid-

THD+N (% OF OUTPUT LEVEL) vs FREQUENCY, 80 kHz LP FILTER

8Ω, 8Ω TAP

20W

1W

THD+N (%)

5

1

0.1

0.01

0.005

10

100

1K

10K

30K

FREQUENCY (Hz)

• (3% THD = SATURATION)

GA-2643-28

ity). The large white tubular 8k resistor mounted on the top of the chassis showed evidence of overheating once I removed the cement wirewound resistors below it. The cement was cracked at one terminal, and both terminals were rusty.

When I stripped back the leads on all three transformers to get fresh copper, I found the copper on all the leads had turned green. I carefully removed the oxide coating down to bare copper and re-tinned all the leads.

LUMPS OF IRON

The design of fixed-frequency power transformers is fairly straightforward. They use 12-mil thick silicon steel laminations for 60Hz or 14-mil for 50/60Hz designs (aerospace 400Hz transformers use 4-mil laminations). The silicon content increases the electrical resistivity of the steel and limits eddy current losses. Carlite insulation is used on the surface of the steel to keep the eddy currents from flowing across the lamination stack.

Output transformer design is an art and science of compromises⁷. An output transformer is used to match the high impedance of the output tubes to the relatively low impedance of the loudspeaker over a wide frequency range. It consists of two electrical windings—a primary with NP turns and a secondary with NS turns—coupled by a magnetic core. The primary winding is divided into two mirror-image sections to minimize the leakage inductance. The secondary may consist of multiple parallel windings that are interleaved within the primary.

Most push-pull output transformers use 14-mil 3% silicon steel E-I laminations, although early designs used as much as 4% silicon content. These thicker laminations are cheaper and easier to stack, but the core losses are greater at high frequency than with thinner laminations. For instance, the core loss for 14-mil 3% silicon steel at a fixed flux density of 3kgauss is only 0.015W/lb at 20Hz, but rises to a whopping 50W/lb at 5kHz.

Fortunately, power amplifiers are operated with as flat a frequency response as possible, and with a fixed output voltage the flux density decreases in direct proportion to frequency. Still, the

core losses increase exponentially with frequency, so the flux density must be kept much lower than that of the power transformer. This, as well as the requirement for low frequency response, requires a lot of iron.

This is the reason that the power transformer, which handles hundreds of watts, is not much bigger than the output transformer. The power transformer is designed for 16kgauss at the AC line frequency, while the output transformer operates with a flux density of only 1.2 kgauss at 1kHz. Extra margin is added for the higher order harmonics that produce exponentially higher core losses.

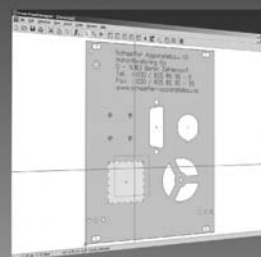
The source and load impedances affect the performance of an output transformer. In the real world these impedances may also change with frequency. The transformer primary winding resistance should be low compared to the source impedance (generally less than 5% of the primary impedance at 1kHz). At low frequencies the leakage reactance and winding capacitance are very low and distortion occurs due to the non-sinusoidal magnetizing current, which is mainly 3rd harmonic.

In order to avoid saturation, the open-circuit inductance should be as large as possible. This requires a large core area that, along with power rating, directly determines the volume of the transformer. Output transformers are also operated at low flux densities, well below the knee of the saturation curve, to minimize their contribution to distortion. High open-circuit inductance also requires a large number of primary turns, because inductance is proportional to the core area and the square of the number of winding turns. However, the more turns in the primary, the greater the winding resistance and distributed capacitance that ends up reducing the high-frequency response.

This high-frequency response is determined by the leakage inductance, winding resistance and capacitance, and the source and load impedances. The resonant frequency defined by the LC leakage inductance and winding capacitance should be as high as possible because the response is relatively flat up to this frequency, and has a second-order rolloff (-12dB/octave) above resonance. Good input and output impedance matching

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is necessary in order to limit distortion and high-frequency peaking. Because the LC circuit introduces phase shift that can eventually reach 180°, feedback amplifiers require careful design in order to avoid peaking, underdamped ringing, or even oscillation. The high-frequency rolloff of the feedback network must be well below that of the output transformer, because feedback cannot be used to extend frequency response.

Ed Dell told me that Scott would all their own transformers. The Scott TRA-8-5 comprises a total of 160 E-I lamination pairs of 14-mil silicon steel with Carlite C10 organic insulation finish (newer transformers might use C5 organic/inorganic finish). I wanted to check the frequency response of the TRA-8-5-1 output transformers to see whether I could extend the amplifier LF response a bit.

For my own information, I measured the winding resistances, the open-circuit primary inductance, the leakage induc-

tance of each winding, and the primary-secondary capacitance. Without knowing the core area, I could not model the transformer core saturation, so I could not really predict the LF response roll-off. The next task was to run some tests, and to measure the actual saturation characteristics.

At no load, the primary exciting current was 1.5mA at 145V RMS 60Hz, driving one half of the total center-tapped primary winding. **A note of caution:** You should not operate vacuum tube amplifiers without a load. The high reflected secondary resistance can react with parasitic capacitance to cause ringing that may impress voltages across the output transformer windings that exceed their voltage rating. Similarly, operating into an output short-circuit could draw too much current from the output tubes.

OUTPUT TRANSFORMER MEASUREMENTS

Figure 26 shows the frequency and phase response of the Scott TRA-8-

5-1 output transformer driving one 8Ω secondary with a 200W power amplifier through a series tap-matching 8Ω resistor. The two primaries are interconnected, and the output of the second transformer has a tap-matching 8Ω load resistor (**Fig. 27**). I measured the transformer current with a Tek A6302 hall-effect current probe and AM 503 current probe amplifier.

The top curve in **Fig. 26** is the frequency response at the three limits of transformer operation. Between 10Hz and 25Hz the power handling capability of the output transformer was limited by LF saturation, which I held to a fixed 3% THD. From 25Hz to 50kHz the transformer was capable of its full 20W output power rating with no noticeable temperature rise. Beyond 50kHz the primary current requirement was held to the 72mA plate current limit I set for the 7189 output tubes.

Just below the “limit” curve at low frequency is the equivalent response of the steep preamp input high-pass filter. The two dashed middle curves represent

FIGURE 29: Scott 222C output transformer 20Hz distortion residual at 3% THD.

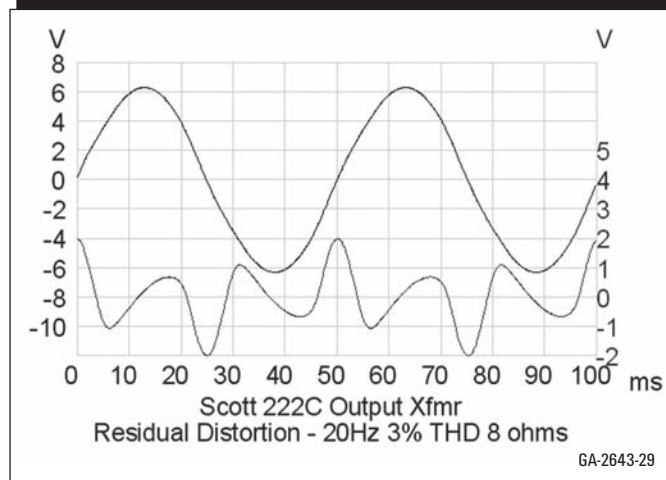


FIGURE 31: Scott 222C output transformer 1kHz spectrum at 1W.

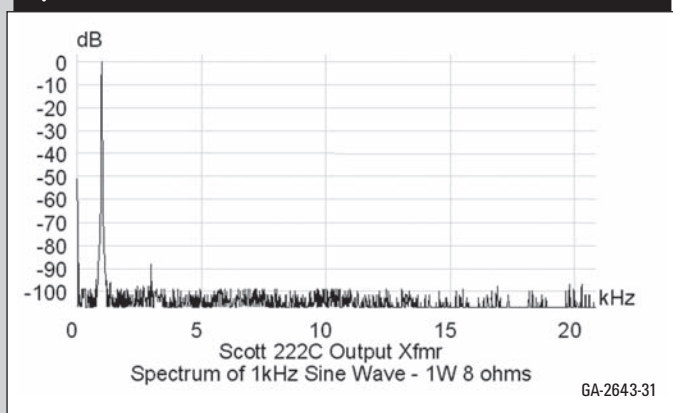


FIGURE 30: Scott 222C output transformer 20Hz spectrum at 3% THD.

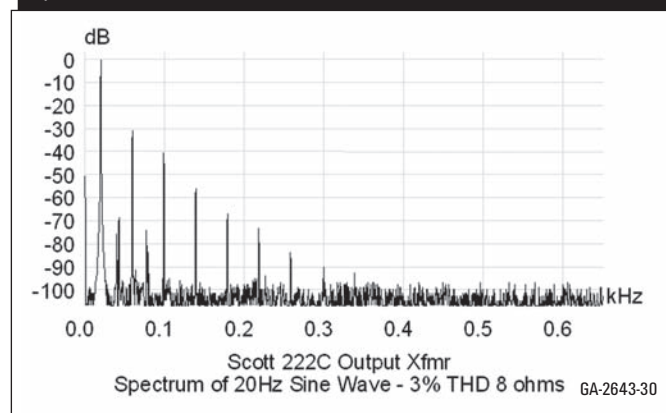
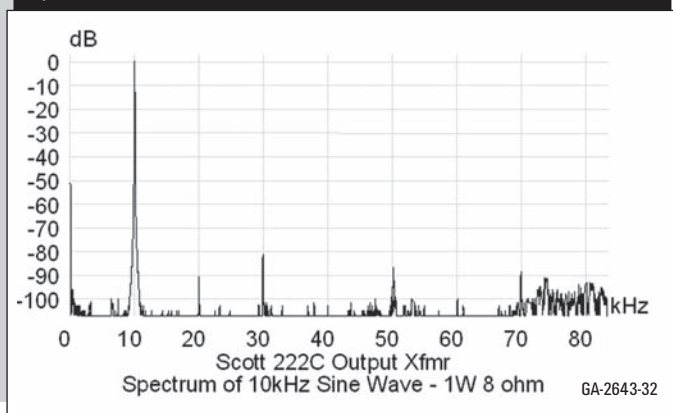


FIGURE 32: Scott 222C output transformer 10kHz spectrum at 1W.



the phase shift between the primary and secondary, first at the "limit" conditions, and then the smoother curve at a fixed 2.83V RMS (1W) into 8Ω. Phase shift in the output transformer does not by itself introduce significant output distortion.

The bottom curve is the transformer frequency response with a fixed 2.83V RMS output into the 8Ω load resistor. Note the peak in response and corresponding changes in phase shift, at about 80kHz due to the increasing winding capacitance. The amplifier response is deliberately rolled off below this frequency to avoid potential instability problems (see **Fig. 10** in Part 1).

Figure 28 shows the THD versus frequency for the output transformer driven by the test setup in **Fig. 27**. Again, I limited the low-frequency output power to the 3% distortion point where the transformer begins to saturate. This graph is limited to 30kHz, but with the LP filter turned off in the distortion test set, the THD began to drop again above 50kHz. This may be due to the increasing high-pass filter effect of the winding capacitance. The THD at 1W is shown as a dashed line.

The output transformer distortion residual for 1W into 8Ω at 20Hz is shown in **Fig. 29**, where I held the distortion to my 3% limit. The upper waveform is the amplifier output signal, and the lower waveform is the monitor output (after the THD test set notch filter), not to scale. This distortion residual signal is predominantly third harmonic as expected. The spectrum of this waveform is shown in **Fig. 30**.

Note that in addition to the smoothly decreasing series of odd harmonics, the 2nd and 4th are also present, albeit at low levels. This indicates a small unbalance in the primary windings of less than 0.03%.

The spectrums for 1kHz and 10kHz at 1W into 8Ω are shown in **Figs. 31** and **32**. At 1kHz the THD+N measures 0.011%, and only the 3rd harmonic is present above the noise floor. In **Fig. 32** there are more odd harmonics present even though the THD+N is still a very low 0.012%. The burst of noise near 75kHz is due to the switching power supply in the computer connected to the digital storage oscilloscope.

The test data for the TRA-8-5-1 output transformer shows it is quite capable of wideband low-distortion operation.

CLEANUP AND REPAINTING

With all the components removed from the chassis, I used car wash powder and water to remove all the dirt, flux, and corrosion from the four aluminum chassis sections. Some of the lettering on the rear panel—already in bad shape—came off with just this mild cleaning. Because I needed to re-letter some areas of the rear panel anyway, I decided to remove all the factory lettering, including the large NOTICE from the top of the front panel informing the owner of how to contact the factory for warranty service. I resorted to steam and a plastic scraper to remove the metal call, then paint thinner to remove the residual glue.

Next I used Meguiars Mag and Aluminum Polish to clean and brighten all the visible areas of bare aluminum. In some cases, I used Bon-Ami first to remove scratches, followed by the Mag and Aluminum Polish. I didn't waste time polishing the inside of the chassis, the front surface of the front chassis (it is covered by the face plate), or the bottom panel.

With the transformers removed, I masked the wires and laminations, and sanded and repainted the transformer end bells and mounting hardware with semi-gloss black enamel. I decided against painting the laminations because at high power the laminations vibrate at twice the electrical frequency, which in my experience eventually causes the paint to flake off. There are special varnishes and epoxies made for this purpose, but they use industrial processes that are beyond the capability of the hobbyist.

Next issue I will detail the changes I made to the schematics and power up the revised unit for listening tests. *aK*

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7. *Electronic Transformers and Circuits*, 3rd Edition, 1988; R. Lee, L. Wilson, C. Carter, Chapter 5 "Amplifier Transformers"; Wiley ISBN 0-471-81976-X.

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BOOK REVIEW

Katz "The Master's Class"

By David J. Weinberg

Regardless of music genre, creating an album involves several technical steps—recording, mixing, and mastering. Renowned mastering engineer Bob Katz (Digital Domain; www.digido.com) has written a book for audio professionals that should be mandatory for all who will ever do any or all of these things; those who design and build hardware and software for music's creation and playback.

Katz maintains that "the mastering studio is the place where experience in the musical art is combined with the science of audio, but the dividing line between art and science is nebulous, so my book constantly tries to integrate the art and the science. [It] is designed to help you learn to make informed decisions on your own; how audio equipment works, and what happens when you turn the knobs."

His book, *Mastering Audio: The Art and the Science* (Focal Press; www.FocalPress.com; 2002; ISBN 0-240-80545-3; \$40) is a masterful tutorial in using science to create and support art.

Of the three steps I mentioned at the beginning, Katz concentrates on mastering, which includes many of the same considerations, techniques, and technologies used in mixing, with added emphasis on assuring optimum performance from the final product.

Roger Nichols says it best in the Foreword: "What places this book head and shoulders above the rest is the attention to useful detail. Instead of hyperbole, the reader can actually put these methods to good use. The descriptions of how to perform a task are augmented with the reason that you should perform the task. [For example:] Not just how downward compressors work, but when and why you would want to use them. Science is meaningless without art."

Nichols suggests that "limiters and compressors should be treated just like firearms. There should be guides for proper use, and classes you must take before you can own one. That class is here in this book. After you read this "audio firearms manual" you will have a much better understanding of the mastering process. You will know when and how to use these tools yourself and when to leave it to the professional."

Katz sprinkles the text with maxims, for example: "Mastering is the art of compromise. . . the first principle of mastering is: Every action affects everything else" and "making good sound is like preparing good food. If you overcook, it loses its taste." A bit facile, perhaps, but the catchy phrases will help novices keep the concepts in mind.

He quotes Glenn Meadows, another mastering expert, who emphasizes the greater importance of the operator than the equipment: "There is no magic silver bullet. There is no one magic anything that will be "best" in all situations. The ability of the operator to determine what it is that needs to be done and pick the best combination of tools is more important than what tools are used."

Katz elaborates: "The truth is that in a typical mastering session, each tool makes only an incremental improvement, and the final result comes from the synergistic totality of the tools working together. In these days of mass-gear-marketing by competitive manufacturers there is too much emphasis on the glitz, fashion and style of the gear rather than on its sound quality and principles of operation. While this book is definitely for gearheads, serious engineers who want to improve their techniques will also find out how their devices function. Audio principles never go out of style, but models of gear will always

fade away."

This text clearly teaches the complexity involved in mastering and how this process can substantially affect an album's musical impact.

Chapter 3. "An Earientation Session: Ear training is really mind training because the appreciation of sound is a learned experience." Katz provides a frequency chart on which he has marked the scale with terms describing various sound affectations, such as "muddy" (~160-225Hz), "nasal" (~750-1.3kHz), "sibilant" (~3-9kHz). He describes a series of ear-training exercises that can be performed easily with minimal equipment, reminding the reader to "take the opportunity to experience the sound of live unamplified music... Surprisingly, some engineer/artists don't know what uncompressed, natural-sounding audio sounds like. While more and more music is created in the control room, I think it's good to learn how to capture natural sound before moving into the abstract. Picasso was a creative genius, but he approached his art systematically, first mastering the natural plastic arts before moving into his cubist period. Similarly, it's good practice to know the real sound of instruments."

A foldout is enclosed of E.J. Quinby's famous 1941 Carnegie Hall "Musical Pitch Relation Chart—Showing Compass of Fundamentals." Over an axis of 109 piano keys (from nine notes below A1 to 12 notes above C8), the frequency of each key is listed, as are the approximate frequency ranges covered by the fundamental notes of singers and orchestral instruments. At the bottom of the chart are musical notation staves, with each C drawn in. There is much more information in this chart; its study is well worth your time.

Chapter 4. Wordlengths and Dither: This chapter presents the best all-around

explanation I've read on dither: what it is, why it is important, and how to use it properly and effectively to maximize sonic quality. Katz tells why it is necessary for DSP used in mixing, editing, and mastering to work with many more bits than the wordlength used in the original recording and the final product. "Each individual decimal place may seem insignificant, but DSPs require repeated precision calculations to perform filtering, equalization, and compression, and the end number may not resemble the right product at all unless adequate precision is maintained. Remember, the more precision, the cleaner your digital audio will sound in the end (up to a reasonable limit)." The spectral graphics in the color plate section show clearly why higher-precision math is required for intermediate signal processing.

Chapter 18. High Sample Rates: Katz considers the requirements and advantages of higher sampling rates, particularly as they benefit the sound of a lower-sample-rate final product, like the CD. Though not placed adjacent to Chapter 4, the subjects of these two chapters are related.

Chapter 5. Decibels for Dummies: Katz is clear in his conviction, well-supported by rational explanations and experiments, that many measurements of equipment performance do not correlate with sonic performance. Level meters with dB scales have been misused since they were invented, especially in the migration from average-reading meters for analog recordings to peak-reading meters for digital recordings. Meters have changed from having a limited, though inaccurate, relationship to loudness, to merely ensuring the recording medium isn't overdriven. One example: "Some mastering engineers deliberately severely clip the signal, and then drop the level slightly, so that the meters will not show any OVERs. This practice, known as SHRED, produces very fatiguing (and potentially boringly similar) recordings." Katz advises that when interconnecting studio equipment, nominal level, clip level, and noise floor need to be balanced to prevent unnecessary loss of sound quality.

Chapter 6. Monitoring: "The major goal of a professional mastering studio is to make subjective judgments as objectively as possible. [A key to this accomplishment is] the intelligent use of an accurate, high-resolution monitoring system. . . The mastering engineer's monitor system is an

audio microscope."

*****Katz's theme: Make the master sound as good as possible on a high quality home audio system, and it will almost always also sound fine in a car, on radio, from boomboxes, and even over the Internet. This is an improvement over an older, mass-market practice, in which some recordings would have their dynamics deliberately altered to optimize the listening experience in a noisy, frequency-limited environment such as a car. With the passing of AM radio as a music source and marketing engine, the older practice is not as useful as it once was.

Chapters 9, 10, and 11. How to Manipulate Dynamic Range for Fun and Profit: "It's not how loud you make it. It's how you make it loud." He explains the effect of various compressors and limiters typically used in attempts to make the recording sound louder, pointing out that the result is often not as effective as intended, and results in unwanted side effects.

At the other end of the level spectrum, "What distinguishes a good noise-reduction job from a bad one?—Good taste. The engineer must continually retain perspective, because the more noise removed, the more noise revealed (noise itself masks other noise below it)! It's like peeling the layers of an onion. If you remove some crackle from the right channel, suddenly you may hear some tics which were not previously audible in the left. In all cases, careful comparison between the source and the processed product is necessary to ensure that the music has not been damaged."

Katz has found that the differences among monitoring systems and their setup have been instrumental in the sound quality disparity found among today's CDs. He is also of the opinion that those recording, mixing, and mastering these CDs have been paying far too much attention to the technology, and not enough to the music: "In the 20th century we concentrated on the medium. In the 21st century, we should concentrate on the message."

Chapters 14 and 15. How to Make Better Recordings in the 21st Century: "Calibrated monitors are the critical tools of the 21st century audio engineer." Katz makes his case for calibrating the mixing and mastering engineers' monitoring systems to a standard level, recommending SMPTE RP 200 ("Relative and Absolute

Sound Pressure Levels for Motion-Picture Multi-channel Sound Systems—Applicable for Analog Photographic Film Audio, Digital Photographic Film Audio and D-Cinema") as the guide. Experiments he and other respected practitioners have conducted, in which level adjustments were made without looking at the meters, consistently resulted in levels that are consistent with the specifications in RP 200. This will temper the widely diverse levels found on recent releases, and help alleviate the variations in bass response that come from mixing/mastering at different levels, since human sensitivity to bass changes dramatically with monitoring level, resulting in different EQ.

The ideal monitoring level is the orthotelephonic reference condition: the playback level is identical to that which would be heard by the listener in the selected location at the "live" event, such as Boston's Symphony Hall seat N22 for a BSO concert, if that is what the producer and recordist want to reproduce or create. To provide an obvious example, a harpsichord is much quieter than a brass band, and each should be monitored, mixed, and EQ'd at its respective orthotelephonic level if the result is to sound the way it does "live." This will aid in maintaining lifelike perceived frequency balance across disparate recordings.

Humans also suffer from listener fatigue, especially at high levels, that results in changes in our perception as we work on a project. This will lead to a change in the sound from the beginning to the end of the album, or possibly even the song. Katz points out that no meters accurately correlate with our hearing, so meter value changes do not match how much we perceive the sound to have changed. "Because of the big difference between typical metering systems and our perception, two pieces of music that measure the same on an SPL or VU meter can have drastically different loudness, depending on many factors including transient and frequency response, and the duration of the sound."

Katz presents his K-system metering and monitoring proposal "that integrates the best concepts of the past with current psychoacoustic knowledge in order to avoid the chaos of the last twenty years." It is based on a metering system calibrated such that 0dB with a mid-frequency sine wave yields a reference loudness from the

monitoring system, and provides a predictable amount of headroom. Using his system should make recording levels and quality more consistent and predictable.

Chapter 16. Analog and Digital Processing: Katz explains “the fallacy of typical weighting curves” such as A or C weighting, which when used for measurements “do not correlate with what our ears tell us.” He suggests considering the F-curve, “a 9th-order curve used by some of the best-sounding dithers.”

“Cheap digital equipment is subject to edgy sounding distortion which can be caused by sharp filters, low sample rates, poor conversion technology, low resolution (short wordlength), poor analog stages, jitter, improper dither, clock leakage in analog stages due to bad circuit board design and many others, such as placing A/D and D/A converters inside the same chassis with motors and spinning heads. It takes superior power supply and shielding design to make an integrated digital tape recorder that sounds good.”

This book points out that while bits are bits, there are great, sound-altering variations among digital-audio hardware and software used in recording, mixing, mastering, and playback. Katz explains these differences, citing why they exist, and how they affect sound.

Chapter 17. How to Achieve Depth and Dimension in Recording, Mixing and Mastering: Katz addresses acoustics, psychoacoustics, plus microphone selection and technique. “I placed this acoustics lesson in the middle of a book on mastering because the creation of wonderful audio masters requires that some basic acoustic principles be understood. As we enter the era of surround recording and reproduction, many mix engineers are repeating their mistakes from two-channel work—panpotting mono instruments to discrete locations, and then adding multiple layers of uncorrelated stereophonic reverb “wash” in a vain and misguided attempt to create space and depth. . . I’ve been making “naturalistic” two-channel recordings for many years taking advantage of room acoustics, but it is also possible to use artificial means to simulate depth. . .”

He prefers to use instrument placement in a room, combined with specific mike techniques, to put the instrument in its proper space, but will also artificially employ the “Haas effect, delays and alteration

of phase, more naturalistic reverberators, and understanding how to unmask via placement.” Another technique is to adjust frequency response to simulate depth, taking advantage of the apparent attenuation of an instrument’s high frequencies with increasing distance.

Microphone placement will have a substantial effect on perception of orchestra depth. For example, if the mike is placed over the orchestra, the sound will have less depth than if it is close to the front of the orchestra. As the mike is moved back in the hall, the sense of depth decreases, as the ratio of distance from the mike to the front versus the back of the orchestra decreases. Katz discusses directivity of instruments, the influence of hall characteristics, various mike techniques including multi-miking, even the influence of the control room. He recommends several commercial recordings to demonstrate various effects.

Chapter 19. Jitter—Separating the Myths from the Mysteries: Katz presents a clear explanation of jitter and its effect on sound quality, including a special section, “Jitter, When it Matters, and When it Doesn’t.”

“There is no jitter on a storage medium.” There is also no jitter in digital processing, since the data is not being clocked, each data value is being arithmetically manipulated irrespective of time.

Katz provides several real-world examples in support of his conclusions, which include the need for careful selection of A/D and D/A converters, the importance of the clock controlling the converters, and how it impacts the sound: “Remember: audio processors process data, not clock. If they hear a difference, it is because a cleaner clock is passed to the D/A converter; but there is no difference in the data being processed;” and “Jitter usually becomes meaningful in a digital mix only during monitoring, when the data is clocked out of a D/A converter.” He also describes various digital audio interfaces, along with several clock systems’ strengths and weaknesses.

He explains why and how measurements conflict with perception; for example: “Traditional audio signal-to-noise ratio measurements have (almost) no relationship to the sound of a converter when it is receiving signal. . . Most signal-to-noise measurements quoted in manuals are therefore irrelevant, and most people have

never heard true 20-bit performance, let alone 24.”

Chapter 21. Education, Education, Education: This is perhaps the most important chapter, as it emphasizes the need for both formal and experiential continuing education—always work toward improving your knowledge, understanding, and skills. Sources of additional information are in **Appendix 10.** Recommended Reading, CDs for Equipment Testing and Ear Training.

CONCLUSION

Mastering Audio’s contents are substantively dense, and of exceptional pragmatic value. Throughout the book Katz explains why some audio truths are self-evident, while others are pure myth. It is important to read the footnotes as an addition to the text, because they contain information that is just as valuable.

Katz’s writing style is direct, with little superfluous content, which results in an engaging read—I didn’t want to stop, except when I felt I had ingested so much that I needed time to process it, within his, and my, context of trying to produce recordings that sound, as much as possible, like live performances (without unwarranted amplification).

His working process and techniques are representative of the best in his profession. We can hope that more individuals involved in recording, mixing, and mastering will adopt and adhere to the principles and procedures that guide his work, so we can enjoy an increasing number of great recordings that sound natural and musical on high quality playback systems, rather than abusively processed for maximum sales to boombox listeners and iPod users.

I could write much more about this book, and still not do it justice. Everyone—those who create music and albums, design music-related hardware and software, and audiophiles—will learn from reading Katz’s *Mastering Audio*, gaining an understanding of the difficulties involved in the task, and the decisions behind the sounds we like and don’t like. **aX**

David J. Weinberg (Tobias Audio, Silver Spring, Maryland; (301) 593-3230; WeinbergDa@cs.com) is an engineering consultant and technical journalist on audio, video, and film technology.

We Interview Peter Bartók

Interviewer Stamler notes that “pioneering recording engineer Peter Bartók was one of the best of the early LP-era engineers, and was involved in some remarkable recordings.”

By Paul J. Stamler

Born into a musical family in Hungary (his father was the renowned composer Béla Bartók, his mother the famed pianist Ditta Pásztory Bartók), Peter Bartók traveled across Europe during World War II to join his parents in America—as it turned out, only a few years before his father’s fatal illness. (The story of his journey, and Peter Bartók’s reminiscences of his father and family, can be found in his book, *My Father*, which I read and found delightful¹.)

Peter Bartók cut records and made tape recordings; much of his work was done using equipment he designed himself, being dissatisfied with commercial-quality gear. In that respect, he qualifies as one of the pioneering audio engineers of the LP era, and I’d venture to say his best work stands up to many of the audiophile recordings released since.

He founded Bartók Records in 1949, which produced critically-acclaimed recordings of his father’s works and other contemporary classical music. But Peter Bartók’s greatest public impact probably came from his behind-the-

scenes involvement with Moses Asch’s Folkways Records; an entire generation grew up learning about American and world traditions from those thick records with blue labels and heavy, double-pocketed jackets containing extensively-notated booklets. Pete Seeger, Woody Guthrie, Leadbelly, the New Lost City Ramblers, ethnographic tapes from every corner of the earth—even the first recordings of avant-garde composer John Cage—passed through the crowded office of Moses Asch, and the cutting amplifiers of Peter Bartók. After Mike Seeger (a founder of the Ramblers) was kind enough to put me in touch with him, it seemed logical to start there.

THE EARLY YEARS AND MOSES ASCH

Can I begin by asking about Folkways?

PB: I used to do a freelance recording service. I had a little recording studio that I made out of my father’s apartment in New York initially, and then I moved into something more appropriate. My business is registered as having started in

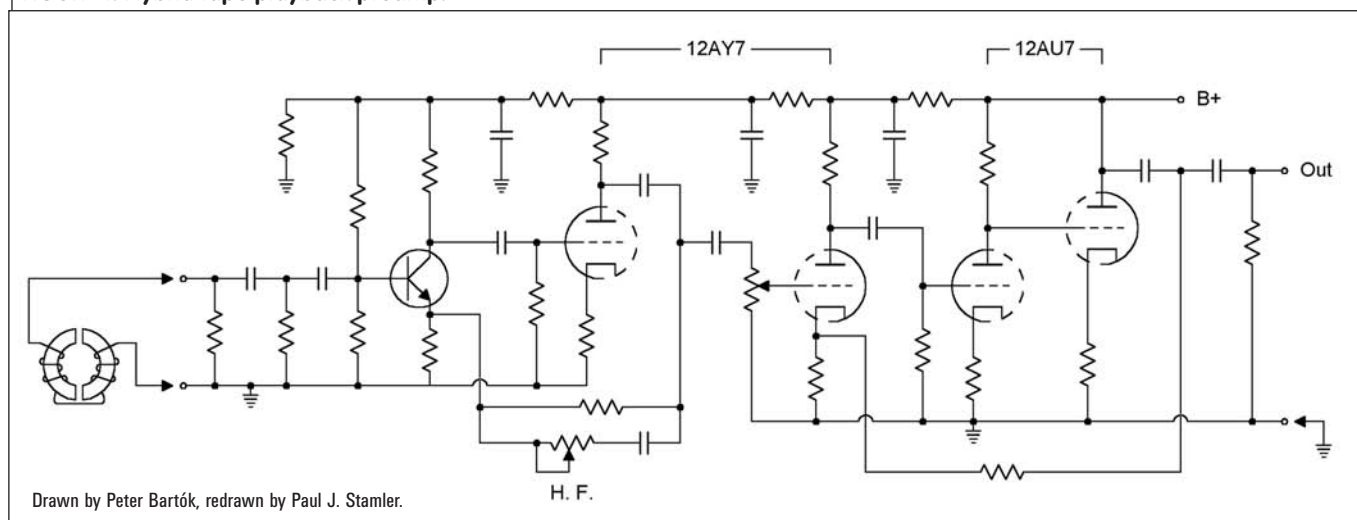
’49, and Folkways was one of my earliest customers for cutting long-playing masters. They brought the tapes, and we transferred them on long-playing lacquer discs, which then went to the factory and were plated.

In my initial work, people came to the studio, played the piano, and recorded. They walked home with a disc. But Folkways brought more interesting folk music records done in the field, which had to be edited sometimes, and then made into long-playing masters. The first record, I think, was a 78 that I made for [Asch].

Just after I started working in this field, my own first record—that I published on my own label—was a 78. I resisted the change [to LP], because,



FIGURE 1: Hybrid tape playback preamp.



Drawn by Peter Bartók, redrawn by Paul J. Stamler.

as a principle, it was a little bit of a downgrading of quality, because of the slower linear velocity of the groove. And then I did like good-quality 78 records; they were capable of wonderful reproduction, once you got rid of the shellac as the pressing material. When the vinyl was introduced, then they were superb.

How late did you continue cutting 78s?

PB: As soon as the LP came out, I stopped; there was no reason to continue making 78s once the LP was introduced. It was possible to get the same good quality, as long as you didn't try to jam too much music on one side. The longer an LP is, the less good the quality.

ON TAPE

How did you first meet Moses Asch?

PB: Well, just exactly who introduced me to him, or who recommended my services to him, I do not know. He brought discs to be copied, and then he brought tapes to be made into masters. I was on 57th St., and he was on 47th St., so it was not difficult for him to get to me.

He was a wonderful man; he was so imaginative and so full of ideas, and he worked day and night. Which to me was not unfamiliar, because my father was that kind of a person also. It was wonderful to work for him; he was such a colorful personality, and it was nice to interrupt the work and go out to lunch, and then come back and resume—in those days I could do that.

And then I got tape equipment; initially I had only disc, because the only kind of tape recording that I knew was not of very good quality. There was a machine called [Brush] Sound Mirror which people had in their homes, but I thought, "Oh, that's not capable of really good reproduction." Finally, a salesman from Ampex demonstrated the Ampex quality, and it made everything very easy. Let us say there was a set of 78 records that we had to copy on an LP; it was easier to put it first on tape than to keep changing the original source.

My first LP record was still done on discs, and during the cutting of the LP master I had to switch the source from one turntable to another, because it was a 15-minute piece which the musicians

could never get right from beginning to end. There were two records, one was good until the middle, and the other was good after the middle, and so the two had to be combined—without using tape. After that I got the tape.

So that was your first machine. Was that the Ampex 300?

PB: That was a 300; I accepted at that time Ampex's claim that 15 [ips] is a good speed for making records. The Sound Mirror, which I didn't like, was going at 7½, and of course the higher the speed, up to a point, the better the quality. And at 15, as I found out later, the quality is really not as good as at 30. Before that time, Ampex marketed a machine which ran at 30. The reduction to 15 [ips] was claimed to be an improvement, but actually it sacrificed some of the quality. So I went back later to 30 [ips] once I realized that 15 was a compromise.

Compromising the sound quality for playing time.

PB: Well, yes, to use less tape material, but a lot of sound was lost, and it had to be equalized excessively to get the same—to get all the frequencies there. Then I received many tapes from Europe that were still 30", and I realized that 30" is the quality that I need.

ROLLING HIS OWN

So we converted the machines by putting a sleeve on the capstan. The Barpex, I used to call it, combined the two names. Ampex didn't like that kind of thing at all; I bought spare parts, replacements, for Ampexes, and instead I put them together into a machine. But the parts could be had, so . . .

The idea was that, when you edit tapes, it's awkward to bend over a factory-made machine. I wanted the machine to be a desk, where I can spend the whole day and work comfortably, where I can put my knees under the desktop, and not get tired. So we decided to build a machine into a desk.

I'm not the only one who did this; people told me that they encountered a similar machine at RCA studios, for editing, [which] went even one step further. They put [in] four turntables for reels, so that they could switch from one reel to the other at any time without having to

remove reels from the turntable. It made editing more easy; we had to keep on taking the reel off and putting the next reel on. RCA could do it by taking the tape from the other reel. [Pause] Editing tapes was a lot of fun. **aX**

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DESIGN NOTES BY PETER BARTÓK

The tape playing amplifier (Fig. 1) is about 30 years old; it was the first time I included a transistor in something I built, but I still felt more comfortable with tubes. What dictated the transistor input stage was the enormous gain required (at low frequencies) and the trouble with microphonics if tubes were used at such a low level.

Followed by only a grounding resistor and an open grid, the transistor with a high resistance load can produce a large gain with very few components. The grid-to-plate capacitance of the following tube is of no consequence, as the gain at high frequencies already is low anyway. I may have included a capacitance-resistor combination in parallel with the transistor's load, so feedback would not be the only factor relied on in obtaining the 6dB/octave drop with frequency rise, but the main equalization determines the frequency response. The variable resistor high frequency levels off the response at a desired point and is, in effect, a variable high-frequency boost that is optional for every tape played. That, and the potentiometer for volume, are the only two controls.

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CORRECTION

The Ed Frias EFE kit (March '06, p.27) obtained from Speaker City used the network that was designed by Ed Frias and *not* the upgraded version designed by GR Research. Measurements of the Ed Frias design and the upgraded GR Research crossover can be seen at www.gr-research.com/edspeaker.htm

MOUSE TUBE AMP

I have a question for Bob McIntyre ("Mighty Mouse: A PP 25L6 Amplifier," April '06, p. 34). First, I like the phase inverter circuit. Also, I thought the use of zeners to set bias was interesting, but I wonder whether there won't still be some degeneration, clipped .6V in one direction and more in the zener direction. Would it, perhaps, be a good idea to have a bypass capacitor across the zeners? Thanks for the good ideas!

*Craig D. Merz
Silver City, N. Mex.*

Bob McIntyre responds:

Thank you for your interest in the Mouse. I enjoyed designing it, and I hope you have fun building it.

The problem I had with the phase inverter was to come up with a simple circuit that would operate well with less than 150V, yet would have low distortion, low impedance, and could easily swing at least 30V. The 6DJ8 was the only tube I could think of that fit these requirements. As it is, I needed to run the tube a little lean in order to fit the current draw within the 10mA maximum rating of the LM334. In turn, I used the LM334 because of its high performance and low insertion loss. I'm glad you like the circuit!

You have a good point regarding the bypassing of the zener bias diodes. I had thought of bypassing them, but decided not to in order to make the circuit as simple

as possible. This does not really become a serious problem, because the output stage never really cuts off, so the diodes stay within their zener region.

The reason that I would consider bypassing the zeners is that their impedance rises with frequency, and their voltage drop varies slightly with current. While I consider these effects negligible, I cannot say absolutely that they will have no audible effect, so bypassing them is a reasonable precaution.

You can determine which of these effects, if any, is dominant, by your choice of capacitors. If the frequency effect is most important, a film capacitor of a few hundred nanofarads will do the trick, while an electrolytic cap of a few hundred microfarads will smooth out the current effect.

SPEAKER SOLUTION

I'm writing with regard to "A Solid-State Single-Ended Power Amp" by Ed Simon in the April issue of *aX*. I can't resist unusual power amps, so as a result I have an 8' rack full of them. I'm building yours and have designed a circuit board so boards will be available for other interested builders.

On page 25 you state "Unfortunately, there does not seem to be a loudspeaker that would use this (1/4W) to produce a practical listening level." Perhaps you should take a look at our model TSMD-2, which has a sensitivity of 97dB. I regularly listen to a pair of them driven from Larry Lisle's "A Minimalist Amplifier Design" (*aX* 3/05), which produce just 1/4W per channel.

*Ron Tipton
TDL Technology, Inc.
www.tdl-tech.com*

Ed Simon responds:

Thanks for the idea. Unfortunately, I blew them up a while ago, and the semiconduc-

Continued on p.62

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Continued from p.59

tors used in them are no longer available. The same part number exists, but the process has changed so they don't sound the same.

VU METER

Without using an integrated-circuit comparator, or an op amp, you should be able to make a bipolar transistor-based five-LED output VU or level display for an audio amp. In this case, though, I want the display to take its input at the preamp level signal as it is applied to a 10k volume pot, so that it indicates the level of input prior to the volume control and so is independent of the volume control. The response should be logarithmic.

I know how to do this with comparators and op amps, but not using *only* bipolar transistors as the log generator and LED drivers. I'd like to see a minimal parts count solution with the applicable math. The power amplifier in question (TEA-2025) has a +12V DC-to-ground supply, an input impedance of 30k, and gain of 45dB, so the applicable preamp level signal is derived from those parameters—having no way to measure it directly—and is fed from the wiper of the 10k volume control. An interesting old-school problem.

*Stephen R. Clanahan
Coalinga, Calif.*

Charles Hansen responds:

The design for even a modest five-step VU (volume unit) meter as described in this letter would have many parts, and is not a trivial design exercise.

In the days before LEDs, IC op amps, and comparators, VU meters were analog. The markings on the meter face did the conversion from a linear audio input signal to the nonlinear VU dB scale. Today you can build a ten-step or more digital VU meter using the National LM3916 dot/bar display driver, whose internal comparator voltage-divider resistors are scaled for VU dB (the 3914 has linear dividers and the 3915 has log dividers). In its simplest form you need to add only two resistors, a capacitor and ten LEDs. Go to www.national.com for the data-sheet.

The discrete five-step VU that Mr. Clanahan requested will require five comparators.

The simplest IC comparators that I knew of were the WE194 designed by Westinghouse Semiconductor (no longer around) and the Fairchild uA710, both designed in the early 1960s, and long out of production. The simplest modern IC comparator is probably the LM339.

The simplest discrete comparator requires a minimum of three transistors: a differential pair to compensate for signal and temperature variations and an output stage to drive the LED. You need seven resistors to provide the bias for the three transistors. You may also need an input buffer stage to be able to drive those five differential pairs without loading the source. The two differential pair transistors should be matched for good accuracy (see PassDIY for a test circuit that can perform the transistor matching).

The voltage-divider resistor values depend on the VU range you wish to display. The LM3916 with a 0-10V input has steps of +3, +2, +1, 0, -1, -3, -5, -7, -10, and -20dB. The National datasheet shows the resistor divider values used. You will also need a stable reference voltage for the divider network, which will take another differential pair amplifier configured as a voltage regulator. You need to select which five of the ten dB steps you want to use, then connect two additional scaling resistors to take the place of the five steps (resistors) you do not use.

Your VU meter can actually be scaled to any desired reference level. The most common reference level in the US is 0VU = +4 dBm, where +4 dBm equals 1.23V RMS measured across a 600Ω load. dBm is shorthand for dB with respect to 1mW, so 0dB is 1mW, or 0.775V across 600Ω. Because Mr. Clanahan is dealing with a much higher impedance, he can scale dB from his input voltage with respect to any DC reference voltage by

$$\text{dBV} = 20 \log (V_{\text{in}}/V_{\text{ref}})$$

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