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PREMIERE ISSUE

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January 2001

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Editorial

The Facts of Life for Magazine Publishing

By Edward T. Dell, Jr.

This first issue of a new magazine seems a good occasion to review for readers just what is required for a healthy periodical. This one, with its pedigree already established, combining as it does 31 years of predecessor publications, is nonetheless totally dependent on three major ingredients—as well as a fourth one.

Readers are all too often seen as passive check writers (or credit-card authorizers) whose only function is to send money. No publisher ever speaks disparagingly of subscriber revenue, but that is not the only—or even the primary—function of a reader.

Readers are, collectively, a kind of community. Their knowledge and experience forms a pool of information, which is ideally enhanced and added to by the publication's content. Inevitably, however, their total knowledge exceeds any amount of content published by the magazine. I have found it uncanny that a very large number of the articles we have published over the years has stimulated some response from one or more readers, which enhance the topic, sometimes adding to it substantially—or possibly shrinking it sometimes when the information is incorrect.

Good magazines flourish when readers share their ideas actively. A publisher always hopes that readers share his or her passion for the subject, not as partisans with religious impulses, but with a desire for the truth of question and an effective understanding of what is usually a search for answers.

Some readers become authors. In truth, authors are only an advanced form of reader. Their interest in the topic has developed experience, and intensified curiosity and probably plenty of experimentation, which is coupled with the desire to share their findings—and their pleasures. Authors are more complex than that. Motives vary a great deal. Seldom can writing for a periodical be for money. Most editors pay at rates that would be illegal under rules for the minimum wage.

Writing for publication has an aura made of components that are not always easy to identify. Fortunately for editors, the sight of one's work on the printed page, disseminated widely, has a lure that continues to attract authors almost as inevitably as bees to flowering blossoms. I have often speculated that writing for publication, either in books or periodicals, has a relation to yearnings for immortality. The objectification of seeing your thoughts in physical form outside yourself seems to have a permanence in the midst of life's flux that is reassuring. All humans, at one time or another, think about mortality and wonder if what they do will make any difference to the world they live in.

Authoring has other appeals. It records a journey, over time building a record. Sometimes it is a stumble which, fully seen, can be updated and corrected. Writing is often a journey rather than a destination. But it is the meat and drink for the editor who is fortunate indeed in his authors—most of the time.

The great preponderance of consumer magazines must have advertisers to exist. Many consumers resent advertisers, mistrust advertising, and have dire thoughts about the oligarchs in big corporations who purchase advertising space or time. I confess to having had similar thoughts about advertisers when I began this enterprise. I believe no avocational magazine can be healthy or successful without advertising.

The money is, of course, very helpful. In a country the size of the U.S.A., the cost of acquiring a new subscriber is very high, often exceeding the price of the first year's subscription. All publishers are happy if a promotion effort response reaches 2%, and many are done for numbers less than that.

For magazines such as this one, however, we need advertisers because they are part of the life blood of a good avocational publication. The goods they offer are vital to the success of the experimenter, the craftsman, and the builder. One of the great developments in magazine technology since the 1960s is the explosion of narrow-interest publications. This development has been a godsend for enthusiasts and smaller entrepreneurs who can get their messages to pre-selected interest groups much more inexpensively.

It often happens that the advertiser in a magazine of this type is a reader who became so interested in the pursuit of good sound that he or she ended up setting up a business. This has happened to many readers of our predecessor titles. This is somewhat risky for the individual, since rarely does a passion for an idea exist in the same body with good business sense. Perhaps that is why many small enterprises begin as partnerships.

Deciding the advertising/editorial content ratio is a critical decision for a magazine's health. The Postal Service limits advertising content in periodicals class publications to 70%. Most publishers keep the ad content between 50 and 60%. Our periodicals have normally kept ad levels at around 40% and often lower. This issue carries 41 pages of ads out of a total of 124. This should make it clear to readers that active response to a periodical's advertisers can increase the editorial content.

These three ingredients, then, make up the proper recipe for a successful, interesting, and enduring magazine. Years ago I began a manuscript for what I hoped would become a book on how to start and publish a magazine. My first absolute requirement for a successful one was a passionate love for the subject.

My passion for good sound carried me along despite long hours of endless work, very slow growth, and even resorting to co-opting members of my young family. The task of producing a magazine is easier today as far as mechanics are concerned, but I think passion is still fundamental. People who publish magazines for the money produce products that readers generally merely tolerate, always hoping to find some kind of satisfaction in the pages somewhere, nibbling on little hors d'oeuvres amid a sea of ads.

I still find magazines whose editor and publisher, I am sure, look forward to coming to work each morning. They stir the imagination, they surprise, they stimulate, they amuse. I am happy to report, somewhat immodestly, that I still enjoy that prospect myself.—E.T.D. ❖

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

JOHN STUART MILL

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Audio News

■ NEW MARSHALL ELECTRONICS MICROPHONE

Marshall Electronics has introduced a new condenser microphone for the home recording enthusiast, the MXL 603. Now you can record piano, guitar, vocals, and other instruments used in the home on a stereo cassette recorder, VCR, DAT, or computer. The mike has a 20mm gold deposited diaphragm with a frequency response of 20–25kHz, operates from a 48V phantom power supply or portable battery box, has a 3 pin XLR for an output connector, and is compatible with standard recorders or computer boards with the addition of various adaptors. Marshall Electronics, (310) 390-6608.

■ SOVTEK 6C45π

Originally designed for military communications and radar for the Russian armed forces, the 6C45π is now available from New Sensor. The 6C45π is a high mu single triode with a low plate resistance and a very high transconductance. The input noise resistance is 100Ω, and the input/output capacitance and the dissipation are nearly 8W, making this tube an ideal candidate for preamp, driver, and signal processing applications. It is also capable of sinking some current, and can be considered a little power amp with gain. New Sensor Corporation, info@sovtek.com, FAX (212) 529-0486.

■ DEHAVILLAND ARIES AMPLIFIER

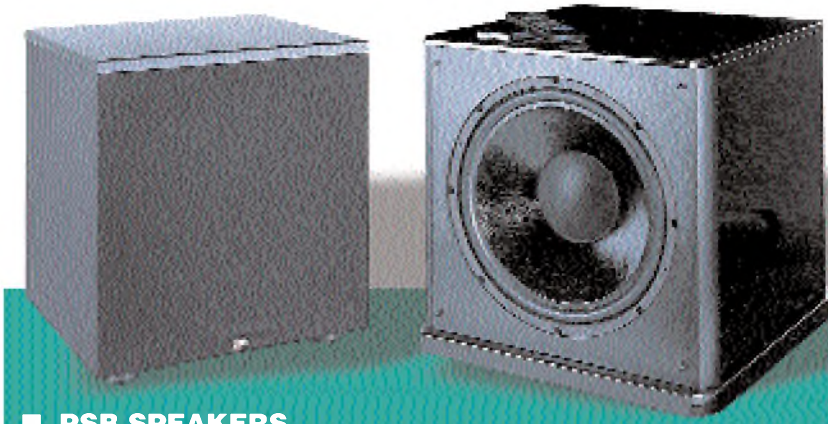
DeHavilland's new Aries Model SE Amplifier is rated at 40W RMS. Using the Svetlana SV-572 directly-heated triode, and Electra-Print transformers, the Aries SE is designed to drive medium-efficiency compact loudspeakers, such as B&W, Sonus Faber, Revel, and Totem. The Aries features hardwired audio circuitry and variable feedback from 0–7dB. Available with optional stepped attenuators, the Aries has sufficient gain to be driven directly from most CD players. DeHavilland Amplifier Company, 1701 Santa Rosa Ave., Santa Rosa, CA 95404, (707) 527-6980, www.dehavillandhifi.com.

■ CEA ONLINE GUIDE

To help shoppers buy, connect, and maintain electronic devices, the Consumer Electronics Association (CEA) is offering an objective, comprehensive source of information. This service, *Switched On, The Complete Guide to Buying and Connecting Consumer Electronics*, is located at http://www.ce.org/switchedon. CEA tapped industry experts, audiophiles, and technology gurus for its information, organizing the specifics by product categories including audio, autosound, computers, home theater, mobile, telecom, and video.

■ DYNAUDIO SUBWOOFER

Dynaudio has introduced the Contour Theater Series Subwoofer, a 210W actively powered Contour Subwoofer employing a proprietary 12″ woofer and offering extremely deep bass response with a high level of speed, accuracy, and precision to intensify the Home Theater experience. Level, phase, and switchable crossover frequency (80Hz, 95Hz, and bypass) control can be adjusted on the subwoofer's back panel, and single-ended inputs are supplied for direct connection from an A/V digital preamplifier with a dedicated subwoofer output. The Contour Sub uses an infrasonic high-pass filter set at 15Hz, 12dB per octave. The unit's low-pass crossover slope is 24dB per octave, and the subwoofer's frequency response is rated at 18Hz–120Hz [±5dB] with a maximum SPL of 110dB at 1m. The cabinet employs a 31mm baffle in a front-ported bass reflex enclosure and is rigid and optimized for the 12″ woofer. Dynaudio, www.DynaudioUSA.com.



■ PSB SPEAKERS

PSB Speakers is renewing its Stratus Series with a new subwoofer, the Stratus SubSonic 7, and two updated center channel systems, the Stratus C5*i* and Stratus C6*i*. The SubSonic 7 employs a 15″ driver coupled to a "BASH" amplifier that delivers 330W RMS, 600W dynamic power, and 1200W dynamic peak-power. The C5*i* and C6*i* both employ a pair of new 1″ neodymium-magnet aluminum-dome tweeters with the distinctive Stratus phase plug, replacing the poly-flare and fabric tweeters in the earlier Stratus centers. In the C5*i*, the new tweeters are coupled with dual 5¼″ polypropylene-cone, rubber-surround woofers, while in the C6*i*, the tweeters are matched with two larger (6½″) polypropylene-cone rubber-surround woofers. Both new centers are available in black ash or high-gloss finish, and the SubSonic 7 is available in black roughcast finish. PSB Speakers, (860) 542-1234, www.pbspeakers.com/media.html.

■ BELL MAGNET COVERS

Jensen Vintage Speakers has re-issued the Bell Magnet Cover for all "R" series ALNICO magnet speakers. The bell is retro-fittable for speakers already in the field and can even be used on original Jensen ALNICO magnet speakers. The Jensen Bell cover is designed to give the speaker the sleek look made famous during the 50s and 60s when Jensen speakers were being used by popular guitarists worldwide. Vintage enthusiasts now have the ability to restore their amplifiers to their original appearance and sound. Jensen Vintage Speakers, CE Distribution, 6221 South Maple Avenue, Tempe, AZ 85283, (425) 744-1053, (480) 755-4712, www.cedist.com.

■ B&K MODEL 885

B&K Precision Corporation announced the addition of the Model 885 Synthesized In-Circuit LCR/ESR Meter to its growing line of test and measurement products. The new, lightweight, battery powered, hand-held unit can be used to test components at frequencies up to 10kHz, and was designed for both component evaluation on the production line and fundamental impedance testing from bench-top applications. The Model 885 offers a large variety of measurement parameters and a wide range of test conditions. It features DC resistance measurement, rechargeable battery/AC power, 0.5% basic accuracy, a large dual LCD display, fully auto/manual selection, and quick response. It comes complete with an instruction manual, battery, SMD Surface Mount Probe, and carry case. B&K Precision Corporation, 1031 Segovia Circle, Placentia, CA 92870-7137, (714) 237-9220, FAX (714) 237-9214, www.bkprecision.com.

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■ LANSONIC™ DAS-750

Lansonic, a division of Digital Voice Systems, Inc., has debuted an MPEG-based audio product, the Lansonic™ DAS-750 "Digital Audio Server." This component is available in a variety of hard drive storage capacities to centrally store music, and combines proven Ethernet computer networking technology with high performance audio circuitry to provide simple and convenient access from virtually any place on a Local Area Network (LAN). The DAS-750, using the MP3 digital format, enables listeners to archive personal album and CD collections all in one location and to store, select, and enjoy thousands of digital quality stereo music selections virtually anywhere on any audio system in a networked home or office. Also, the system's multi-user network capability allows several people to simultaneously listen to different music in different rooms. Digital Voice Systems, One Van de Graaff Drive, Burlington, MA 01803, (781) 270-4546, FAX (781) 270-0166, www.lansonic.com.

A 30W Push-Pull 3CX300A1 Monoblock

This article was originally published in the September 1999 issue of Japan's high-end tube magazine, *MJ Audio Technology*.

BY SATORU KOBAYASHI

This article describes a Svetlana 3CX300A1 push-pull monaural amplifier I built, using the experience gained in fashioning the 3CX300A1 single-ended stereo amplifier first described in *MJ Audio Technology*'s September 1999 issue. This push-pull amplifier features a well-balanced voltage-driver circuit, using a differential amplifier and a cathode follower to drive a pair of 3CX300A1s with low impedance.

The maximum output power is 36W at 3% harmonic distortion, using a 450V plate-supply voltage. The sound of this amplifier is very strong and dynamic, giving a larger presence in vocal sounds than my single-ended amplifier[1] previously mentioned.

PHOTO 1: The completed amplifier.

DESIGN GOALS

- Drive the pair of 3CX300A1s in Class A1, for a predicted maximum output of 30–40W with lower distortion than a comparable single-ended amplifier;
- Use a Plitron toroidal output transformer to yield a power bandwidth of over 100kHz;
- The amplifier will be monaural, to limit weight and chassis size;
- No negative loop feedback will be used, honoring the policy of Menno van der Veen, the designer of Plitron toroidal transformers;
- Use a power MOSFET-regulated ripple filter to save cost and weight (by replacing a filter choke);
- Obtain low ripple voltage by using a three-terminal solid-state voltage regulator for the negative grid bias of the final tubes;
- Use the Tube CAD simulator by Glassware to design the voltage driver for optimum circuit performance;
- Build the power supply circuit onto a PCB to avoid hum by providing star grounding; and
- For easy assembly and maintenance, use PTP terminal boards from International Audio Group to build the entire circuit.

POWER SUPPLY

I used Plitron's toroidal power transformer, model 754709. Even though this is Canadian made, it provides a 100V AC winding for use in Japan. The 340V AC-0.7A winding goes to a bridge diode to rectify, then to a power MOSFET ripple filter stabilized with three 150V zener diodes cascaded to generate 450V DC. The resulting ripple voltage was only 14mV.

The 40V-0.1A AC winding also goes to a bridge circuit. A three-terminal voltage regulator minimizes ripple voltage to provide fixed bias grid supply for the final tubes.

A 470µF 63V electrolytic capacitor bypasses the lower grid resistors of the final tubes, thus decreasing the ripple voltage to only 0.3mV. The 6.3V-6.8A AC winding drives the heaters of all vacuum tubes without any special circuitry—no DC heater power was needed.

Approximately 8V DC is available after rectifying the 6.3V AC. This drives a cooling fan to supply forced air to the final tubes. To reduce fan speed and noise, the fan runs on a lower voltage than its rated 12V.

FINAL STAGE

SUPPLY VOLTAGE AND CURRENT SUPPLY

First of all, you must define the supply voltage. I determined that 450V is a good fit for the available electrolytic capacitors made by Nichicon, Elna, and so on. This voltage allows capacitor overrating for extra safety and longevity.

Once you've decided the supply voltage, you must calculate the total current consumption. For my single 3CX300A1 amplifier mentioned previously, I defined an idle current of 120mA, with a 450V plate supply assumed. So, for push-pull operation, you need 240mA (120mA × 2). Since the voltage driver circuit consumes another 10 to 20mA, the total current is 250–260mA at idle.

3CX300A1 TUBE OPERATING POINT

It is necessary to fix the operating point by assuming the maximum plate current. B.J. Thomson has described the operating

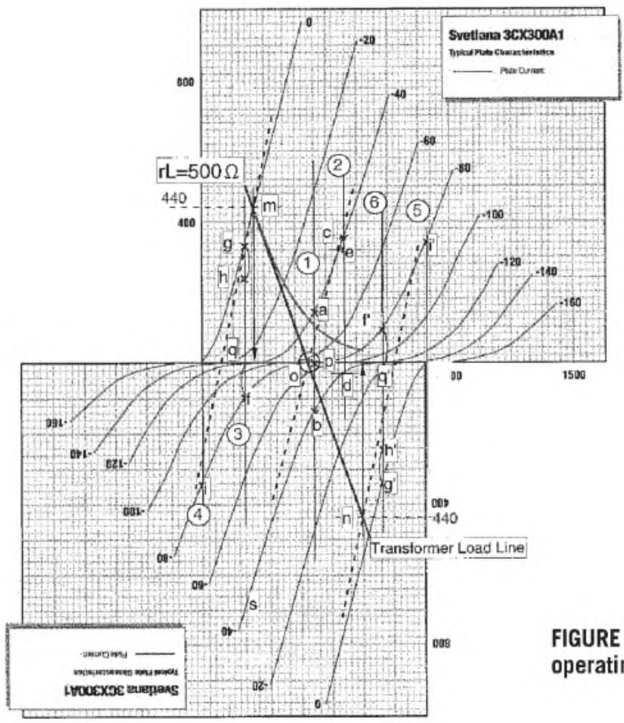

FIGURE 1: 3CX300A1 operating curve.

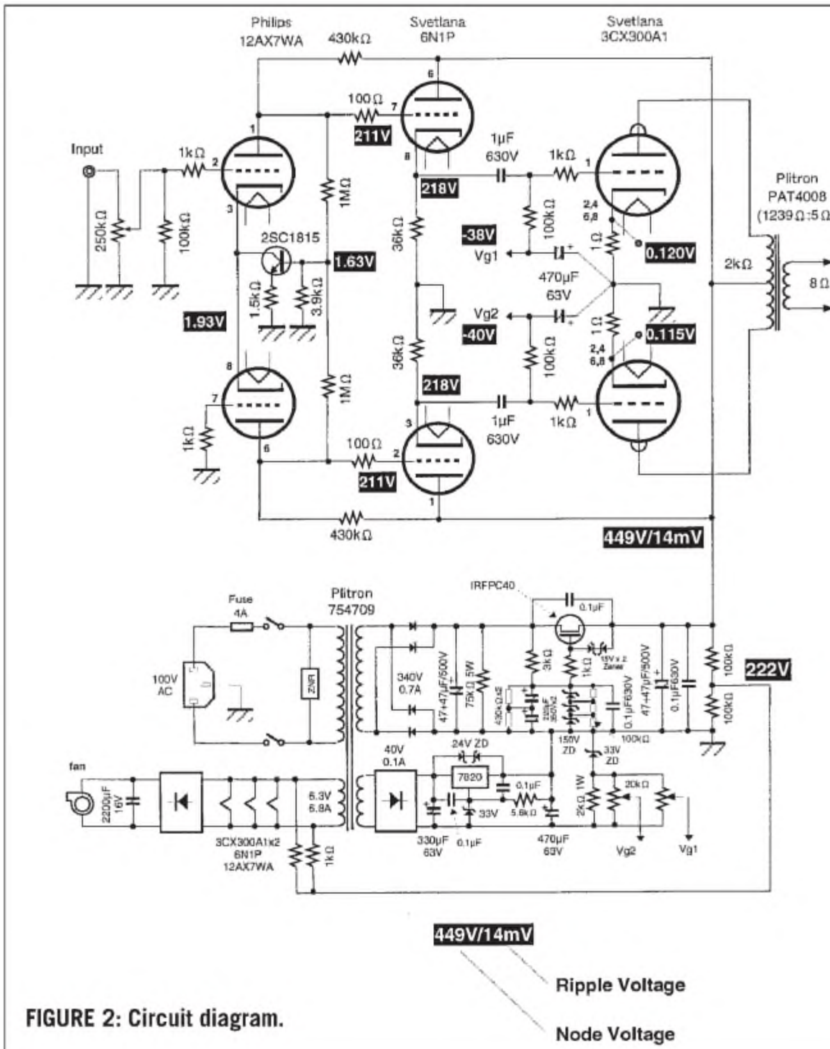

FIGURE 2: Circuit diagram.

condition in his article on graphical analysis for push-pull amps,[2] using the plate curves of the chosen vacuum tubes.

To get this done, I first of all downloaded the Adobe Acrobat (.pdf) file of the 3CX300A1's data from http://www.svetlana.com. (It is very convenient that Svetlana provides this data at their home page.) Using the .pdf file, I then combined the plate operating curves for push-pull operation (*Fig. 1*).

In this graph, the grid voltage of the final tube swings between 0V and −84V from the center idle point of −42V. The plate current is at its maximum value when the grid voltage swings to 0V, at the point of intersection with the output transformer's loadline. Thus, the plate load impedance defines the maximum output power.

DEFINING THE LOAD IMPEDANCE

Svetlana recommends a 1.9kΩ plate-to-plate load for a 3CX300A1 push-pull amplifier. At the design stage, I was familiar with the PAT4008 toroidal transformer from Plitron's home page at http://www.plitron.com. This particular transformer provides a 1239Ω load impedance with a nominal 5Ω secondary, so when an 8Ω load is connected to this tap, the apparent primary impedance is 1k98Ω. This fits in excellently with the recommended loading condition.

It is known that the loadline fits onto the operating curve with a slope of one-quarter of the transformer impedance. Thus you can draw the 500Ω-load impedance line (line m-o-n) over the plate curves, passing through the point O (*Fig. 1*) at 0mA plate current. Then the reading of plate current at the point m (n) is the maximum, e.g., I_{max} = 440mA. The maximum output power is determined by the following formula: $P_{omax} = (I_{max}/2)^2 \times r_L$, giving a value of 48.4W ($(0.44 \div \sqrt{2})^2 \times 500$).

CALCULATING AVERAGE PLATE CURRENT OF FINAL TUBES

Formula 25 shown on page 589 of reference 1, $I_b = I_{bo} + (I_{max} + I_{min} - 2 \times I_{bo})/4$, gives the average plate current under Class A1 operation. Since the plate-current characteristic shows 0mA for I_{min}, this gives I_{max} = 440mA. And if I_{bo} = 120mA, the average current is 170mA per tube. So the total average current of this amplifier is 350-360mA(170 × 2 + 10-20). The power-transformer current capacity of 0.7A AC, with a full-wave bridge rectifier, meets this requirement under the formula $I_{AC} = 1.8 \times I_{DC}$, where I_{DCmax} of the transformer is 389mA (0.7A ÷ 1.8).

MAXIMUM REQUIRED DRIVING VOLTAGE FOR THE FINAL TUBES

It is good to know the maximum driving voltage needed to determine the voltage-driver circuit gain. Since the 42V bias voltage was defined by the idle current, this value forces the peak driving value under Class-A operation while avoiding positive drive of the final tubes. Thus the maximum driving voltage is 29.7V RMS ($42/\sqrt{2}$).

VOLTAGE-DRIVER CIRCUIT AND GAIN

I used the Tube CAD software simulator to determine the parameter values of the driver circuit that give stable operation. In general, most existing tube-amplifier

TABLE 1
SIMULATION RESULT OF DIFFERENTIAL AMPLIFIER USING TUBE MODEL 12AX7

PARAMETERS	B+ = 450V	B+ = 350V
Plate resistance (Ω)	430k	250k
Cathode resistance (Ω)	1.8k	1.5k
Load resistance (Ω)	2.2M	2.2M
Plate current (mA)	1	1.2
Gain (dB)	85.1 (38.6)	78.5 (37.9)
Common-mode rejection ratio (dB)	−3.11	−3.68
Ripple rejection ratio (dB)	−5.32	−4.16
Plate-cathode voltage (V)	233	198
Saturated input voltage (V)	2.11	1.72
Maximum output voltage(V)	−182/+182	−137/+137
Grid-bias voltage (V)	−2.13	−1.76
Input impedance (Ω)	53.4k	53.4k
Output impedance (Ω)	206k	137k
Lower frequency @ −3dB(Hz)	0.07	0.07
Upper frequency @ −3dB(Hz)	>1M	>1M

TABLE 2
SIMULATION RESULT OF CATHODE FOLLOWER USING TUBE MODELS 6N1P (MODIFIED 6DJ8) AND 6FQ7

PARAMETERS	6N1P	6FQ7
Plate voltage (V)	450	450
Cathode resistance (Ω)	36k	36k
Load resistance (Ω)	100k	100k
Plate current (mA)	5.75	5.8
Gain (dB)	0.97/−0.3	0.94/−0.5
Ripple rejection ratio (dB)	−30.9	−27
Plate-cathode voltage (V)	243	231
Saturated input voltage (V)	147	144
Maximum output voltage(V)	−142/+142	−135/+135
Grid-bias voltage (V)	−6.71	−8.34
Input impedance (Ω)	5.68M	2.15M
Output impedance (Ω)	90.6	379
Lower frequency @ −3dB(Hz)	1.59	1.59
Upper frequency @ −3dB(Hz)	>1M	>1M

TABLE 3, PARTS LIST
AMPLIFIER, PER CHANNEL

PART	DESCRIPTION	NUMBER
Vacuum tube	3CX300A1 Svetlana	2
Plate clip	AC-1 Svetlana	2
Vacuum tube	12AX7WA Philips	1
Vacuum tube	6N1P Svetlana	1
Transistor	2SC1815 Toshiba	1
Power transformer	Plitron 754709	1
Output transformer	Plitron PAT-4008	1
Resistor	1Ω 1W	2
Resistor	100Ω ½W	2
Resistor	1kΩ ½W	5
Resistor	1.5kΩ ½W	1
Resistor	3.9kΩ ½W	1
Resistor	36kΩ 3W	2
Resistor	100kΩ ½W	1
Resistor	430kΩ 1W	2
Resistor	1MΩ ½W	2
Potentiometer	250kΩ Audio taper, panel mount, COSMOS	1
Capacitor	1μF 630V Angela Instruments	2
PTP terminal board	Custom made by International Audio Group	2
POWER SUPPLY		
Power MOSFET	IRFPC40 VGS = 600V, Id = 6.8A, Pd = 150W, RDS = 1.2Ω Shin-Dengen	1
Three-terminal regulator	7820 (20V1A)	1
Zener diode	Z6150, 150V 2W Ishizuka Denshi	3
Zener diode	33V 1W Toshiba	2
Zener diode	15V 1W NEC RD-15F	2
Silicon diode	1000V 1A Schottky type, RG4C Shin-Dengen	4
Bridge diode	100V 1A Shin-Dengen	1
Resistor	1kΩ ½W	3
Resistor	2kΩ 1W	1
Resistor	3kΩ 2W	1
Resistor	5.6kΩ ½W	1
Resistor	75kΩ 5W	1
Resistor	100kΩ ½W	2
Resistor	430kΩ 1W	2
Capacitor	0.1μF 50V Ceramic	2
Capacitor	0.1μF 630V Ceramic	1
Electrolytic capacitor	330μF 63V Nichicon	1
Electrolytic capacitor	470μF 63V Nichicon	3
Electrolytic capacitor	220μF 350V Nichicon	2
Electrolytic capacitor	47μF + 47μF 500V LCR	2
ZNR	14K431U Matsushita	1

PART	DESCRIPTION	NUMBER
Heatsink	to fit 7820 regulator IC	1
Heatsink	Mizutani, to fit IRFPC40 MOSFET	1
Pin terminal	Teflon insulator	10–20
Printed circuit board	100 × 75mm, epoxy glass, 1.6mm thick	2
CHASSIS		
Top plate	400 × 200mm stainless steel, t = 1.6mm. Custom made by San-Ei Musen	1
Safety cover	65 × 65 × 75mm, stainless steel, t = 1.2mm. Custom made by San-Ei Musen	2
Chassis	400 × 200 × 60mm, homebrew	1
Wooden Side panel	430 × 67 × 15mm, 210 × 67 × 20mm, 15mm thick, oil-stained, oil-finished	1
Socket	SK-2A, Svetlana	2
Socket, 9-pin minitube	QQQ made	2
RCA jack, panel mount	Supertron	1
AC power entry module	Corcom, made in Mexico	1
Speaker terminal	Black & red, San-Ei Musen	1
Metal knob	San Ei Musen	1
Pilot lamp	110V red, neon, Sato Musen	1
Cooling fan	109R0812T4B03, 12V 0.19A, Sanyo	1
Electrolytic capacitor	16V 2200μF	1
Bridge Diode	100V 1A	1
Metal feet	IAG	4
Stainless mesh	120 × 120mm, S × L	1
Brass spacer	15mm long 3mm screw	8
5/16″ bolt & nut	90mm long	1
6mm bolt	75mm long	2
3mm bolt & nut	35mm long	20
4mm bolt & nut	35mm long	4
Hookup wire	as needed	as needed
Teflon sheet	73 × 150mm 0.5mm thick, Small Parts Inc.	2
Pin terminal	San Ei Musen	as needed

MEASUREMENT EQUIPMENT

Audio analyzer, HP-334A
Audio generator, Kenwood AG-204D
8Ω dummy load, 50W, homebrew 2ch.
Oscilloscope, HP-1746A
600Ω attenuator, HP-4437A
AC voltmeter, HP-403B
Digital multimeter, Fluke 8020A

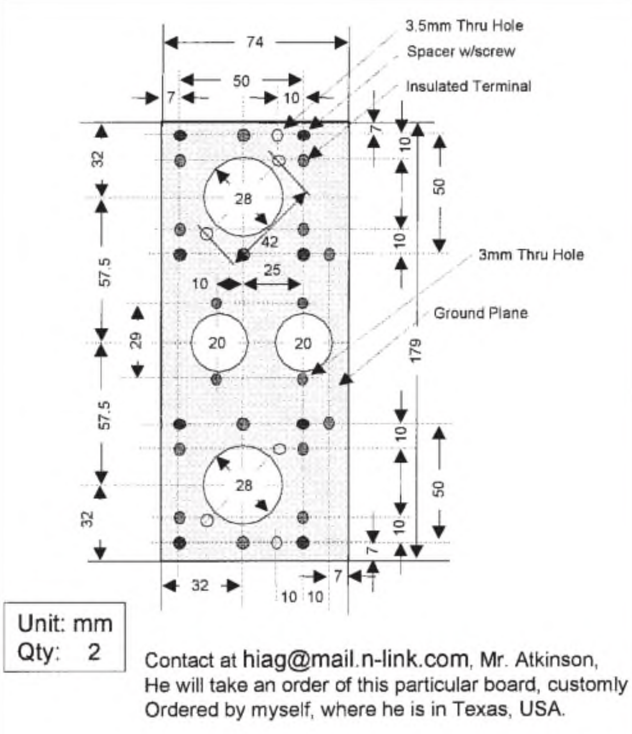

FIGURE 3: PTP terminal board.

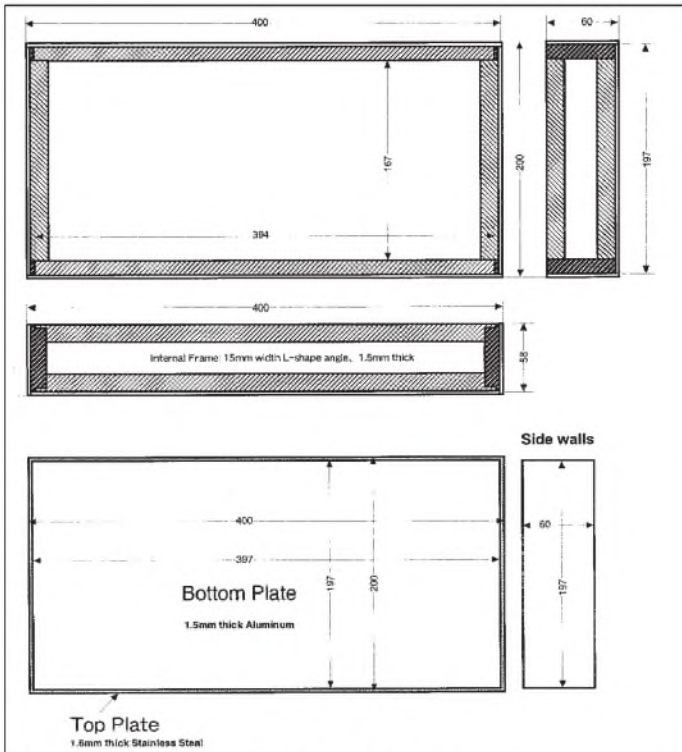

FIGURE 4: Chassis internal frame structure.

designs need a minimum input voltage of 0.5-1.0V RMS to get maximum output power. So the voltage-driver circuit must provide voltage gain between 30(=29.7/1) and 60(29.7/0.5).

CHOOSING THE DRIVER CIRCUIT AND VOLTAGE GAIN

I preferred to use cathode followers to drive the power tubes, since I already knew their good performance. They provide fairly low output impedance, but with a gain of less than 1. Thus the first-stage amplifier needs to amplify the input signal of about 0.5 to 1V RMS with a gain between 35 and 40 (30/0.9), then send the amplified signal to the cathode-follower stage, driving the final stage with its pair of 3CX300A1s. The circuit must have a phase-splitting feature for proper push-pull drive.

The first stage is a differential amplifier, which generates complementary signals to drive a push-pull circuit easily. A dual-triode tube is ideal for this circuit. When the other complementary input is grounded, the total gain is approximately 50% less, so the circuit can work as the phase splitter to provide push-pull drive.

First of all, I simulated this differential circuit using a 12AX7, with its high mu of 100, and with a plate supply of 350V, then 450V. The result showed a gain of 78 and 85, respectively. Once the unused input of the differential amplifier is grounded, the total gain would be 35 to 40. Thus, the circuit will meet the requirement for building a phase splitter (*Table 1*).

Once it was defined, I modified this differential amplifier by using a small signal transistor, such as the 2SC1815, as an active current source, avoiding the need for a negative power supply. Also, this circuit uses the feedback loop from the differential outputs to control the gain.[3] The power-supply voltage is the same 450V as for the final stage, simplifying the internal wiring and assembly.

The cathode-follower circuit uses Svetlana's 6N1P, which has a higher gain than the usual choices, the 6CG7/6FQ7 or 12AU7. This gives a higher voltage gain, closer to 1. The 6N1P also has lower plate resistance, so it saturates the output stage more easily than do the other tubes. The simulation shows a lower output impedance (90Ω) and higher gain (0.97) than would be obtained with a 6FQ7 (*Table 2*).

The final circuit schematic (*Fig. 2*) was

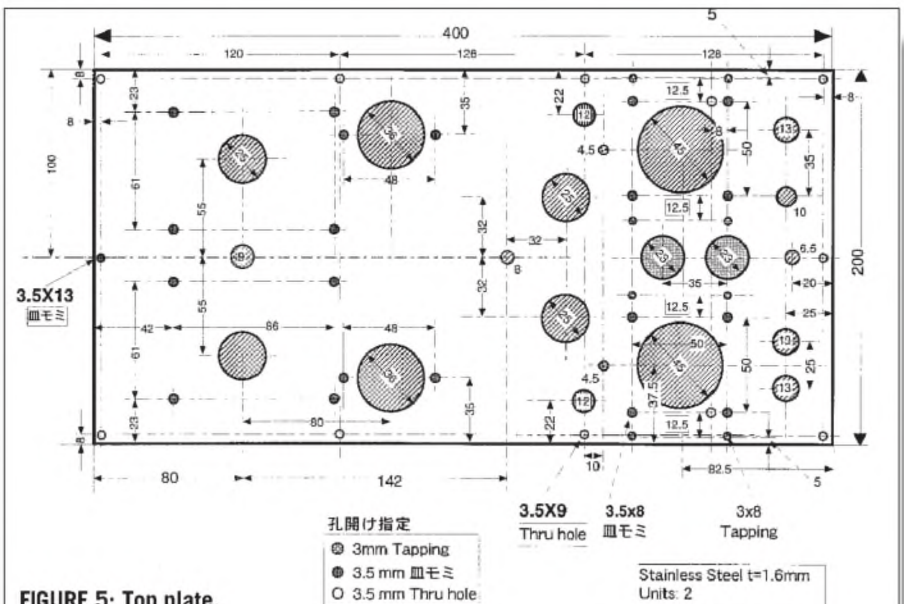

FIGURE 5: Top plate.

completed after the foregoing analysis. The circuit features a differential amplifier using an active current source made with a small signal transistor, and a cathode-follower circuit to drive the outputs. The fixed-bias circuit brings the –42V bias to the grids of the final tube pair.

COLLECTING THE PARTS

With regard to the parts (*Table 3*), I prefer to use Plitron's toroidal power transformer, since it features high efficiency, low waste-heat dissipation, and low leakage flux. I also like the wide frequency response provided by the toroidal output transformer, as well as its good damping and good phase characteristics.

The final tubes are a pair of Svetlana-made 3CX300A1s, with their unusual shape. Since the 3CX300A1 was originally designed for the high-voltage regulators of Russian radar systems, the maximum plate voltage is a very high 1k8V, and the tube structure itself is, according to Svetlana, very rigid, because of the shape of the ceramic and metal parts. Thus, a final tube-protection circuit using a timer relay is not implemented.

Svetlana's 6N1P is roughly equivalent to the 6DJ8, although its maximum plate voltage is 250V, whereas the original 6DJ8 is only 130V. The 12AU7, 6189W, and 6922 are also similar to the 6N1P, but the output impedances of cathode followers made with those tubes appear to be higher than that of the 6N1P. The first stage 12AX7A is a NOS tube by Philips ECG.

The PTP terminal boards, made by International Audio Group (IAG) in Texas, are also used to obtain simplified wiring and easy maintenance. The board is 3.2mm-thick, copper-clad epoxy, featuring a number of turret posts swaged anyplace on the board, as specified by the customer. This structure provides some convenience for simplifying the wiring assembly, as well as shortening the assembly time.

I have suggested to Mr. Atkinson at IAG that they use a copper-clad board instead of a plain board, since I needed a ground plane for easier ground wiring. He accepted this instantly, and the result minimizes hum without using a star-grounding wiring scheme. Thus I love this board, and wish to act as an evangelist. I suggest that audiophiles try to use the IAG boards at least once. They are really neat!

If you choose to obtain this board, I suggest you e-mail (hiag@mail.n-link.com) to IAG a pin-placement drawing in Microsoft PowerPoint file (*Fig. 3*). IAG prefers a clear computer-graphic file when programming its machine, allowing the quickest setup. Even from Japan, I was able to get prompt service. The board came to me in ten days via express mail from Texas. If you would like to use my own board design, you might specify that the board must be identical to mine.

PHOTO 2: Removing a tube cover reveals the internal arrangement of the tubes.

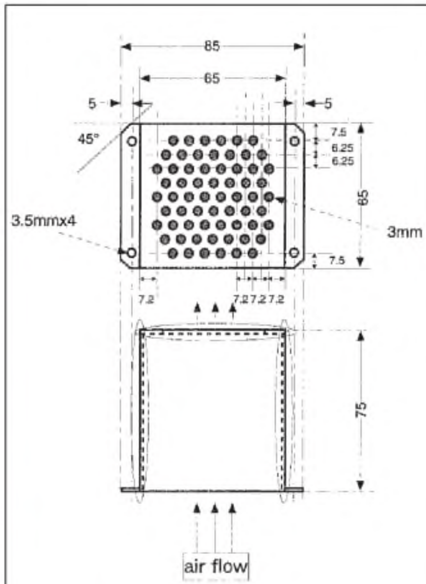

FIGURE 6: Shock proof cover.

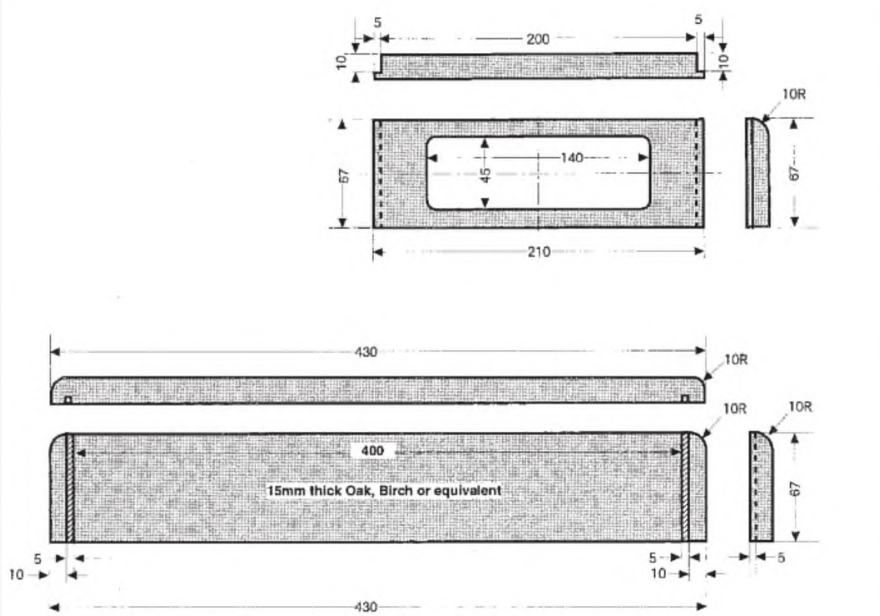

FIGURE 7: Wooden side panel.

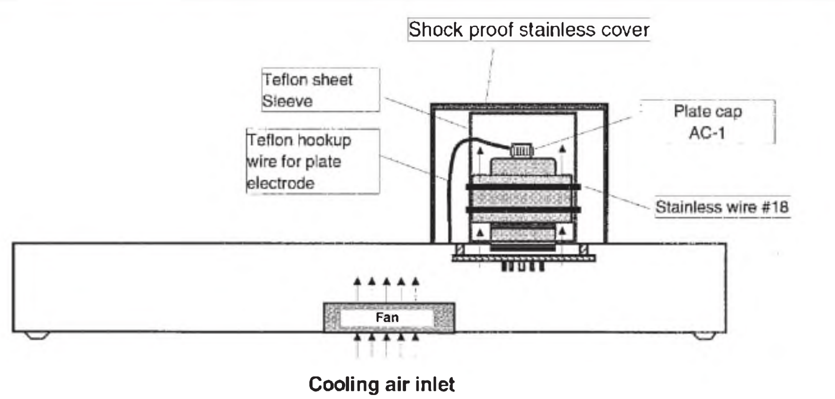

FIGURE 8: Cross section of final tube installation.

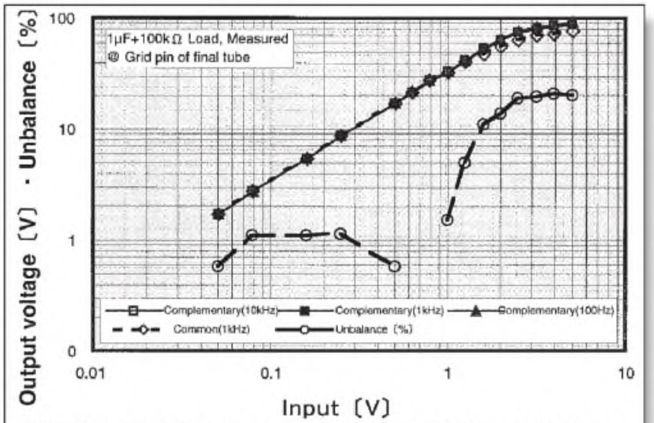

FIGURE 9: Input versus output characteristic of voltage driver.

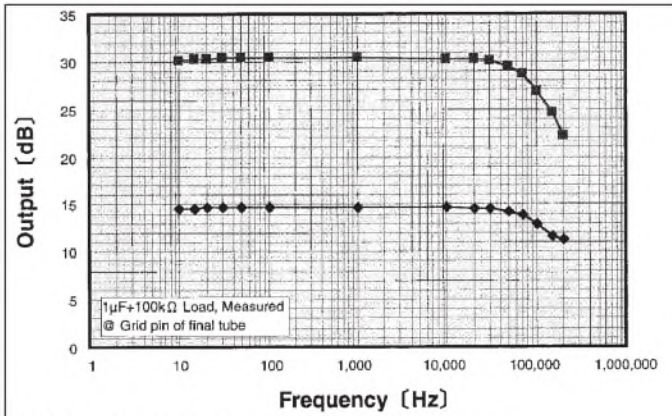

FIGURE 11: Frequency response of voltage driver.

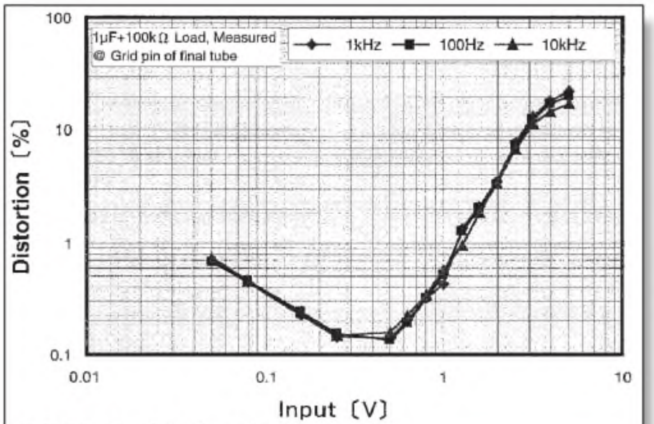

FIGURE 10: Distortion of voltage driver.

PHOTO 3: A forced-air cooling fan was installed at the center of the bottom lid with a stainless mesh.

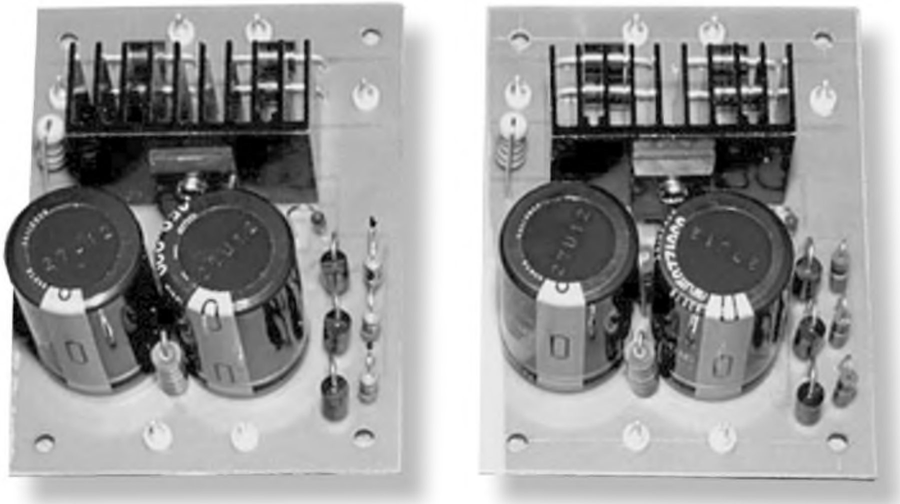

PHOTO 4: The MOSFET and Zener Diode Ripple Filter.

ASSEMBLY

I built the case myself, using pop rivets and other tools (except for the stainless top plate and the protective cover, which were specially made by San-Ei Musen in Akihabara). After assembly, I fixed a 15mm-thick wooden plate, oil-stained and oil-finished, to the sidewall of the case, to give a manufactured appearance (*Figs. 4, 5, 6*, and *7*).

Owing to the professional appearance, the case looks as though it were made by a major manufacturer. I oil-finished the wooden cover three times after oil staining, so that its surface became as smooth as a furniture finish.

I mounted the power-supply board on the internal chassis with a 15mm offset using standoffs. Since the toroidal transformer requires only three small holes to fix it to the top plate, this leaves a lot of room underneath.

Some might say it is very hard to use a ceramic tube like the 3CX300A1 in an audio amplifier, since the tube needs forced-air cooling because of its unique design. As I found, it became very easy to mount these tubes, thanks to the PTP terminal board.

You mount all sockets on top of the board, and then fix the board underneath the top plate with 1cm-tall standoffs. A 45mm-diameter hole on the top plate provides enough room around each 3CX300A1, so that cooling air comes from the fan installed on the bottom plate, passing only through the tube fins (*Fig. 8*).

To improve cooling efficiency, I put a 0.5mm-thick Teflon® sleeve around the tube's fins, winding it firmly with #18 stainless wire. The size of Teflon sheet is approximately 73 × 150mm, so that it covers the sides of the fins and reaches to the same level as the stainless top plate. Thus, no cooling air goes through the gap between the sleeve and the fin, forcing all the air through the top of the tube like a chimney. Over this setup, I used a custom-designed stainless hood to protect users against electrical shock and heat (*Fig. 6*).

The fan generates the forced air. It is installed over the bottom plate, located at the center, and is driven by 7 to 8V DC derived from the 6.3V AC line for the tube heaters in order to reduce fan noise. A stainless mesh filter is placed over the bottom plate. The structure of the chassis is airtight, so that all the forced air exits through the tube fins.

ADJUSTMENT AND MEASUREMENT

First of all, double-check the internal wiring carefully before inserting the tubes or powering on. Then apply AC power and check the plate-supply voltage with a digital multimeter. It is OK if the plate supply is around 450V, the heater voltage is 6.3V or so, and the fan is working correctly. (Place your palm over either air outlet on the top cover. The air-flow rate is very low, much less than a hair dryer. The tubes are not running at their full power, so this is adequate airflow.) Then adjust the grid-bias trimpot on the top plate so that the grid-bias voltage (measured at the test jack) is roughly –50V. Then temporarily turn off the power switch.

After this initial setup, insert all tubes and turn on the power switch. Measure the voltage drop across the 1Ω resistor at the cathode pin of each 3CX300A1, and adjust the grid-bias trimmer so the voltage drop becomes approximately 120mV. Wait for about ten minutes to allow the ceramic tubes to warm-up, since Svetlana says this tube needs several minutes of warm up time. During this period, the cathode-voltage drop is unstable.

Again, measure the voltage drop across the 1Ω resistor at the cathode pin of each 3CX300A1; it should still be around 120mV. Then observe the stability of the reading.

A 100Hz square-wave test is all I did for the final adjustment, although an audio signal generator could give more detail and a more precise adjustment. An example is the 100Hz square-wave test that van der Veen has developed.

An oscilloscope allows precise adjustment. Feed the amp with a 100Hz square wave, then observe the square wave at the output, using an 8Ω dummy load. Adjust the grid-bias volume control so that the output square wave is a perfect rectangle. Careful adjustment according to this scheme results in minimal imbalance of the idle current, usually less than 1mA when using a matched pair tube.

Originally I used a nonmatched pair of 3CX300A1s, since Svetlana does not supply matched pairs at this time. The imbalance of the cathode voltages between the tubes becomes maximum when the output power is at clipping, which can be about 5 to 10mA. This can generate acoustic sound, because the output transformer's toroidal core resonates mechanically at 100Hz. A crazy but workable method for adjusting tube balance suggests itself: adjust the grid-bias volume control so that this audible resonance of the transformer core is minimized.

After adjustment by this method, the idle current of both tubes was 115mA and 120mA, respectively.

OTHER INFORMATION

I finished the adjustment and measurement on the next day. Prior to the measurement, I needed to warm up one amplifier, so I turned the power on. A loud buzzing sound came out of the speaker for a short time.

I had no idea why this happened. To

PHOTO 5: Close-up of tube installation.

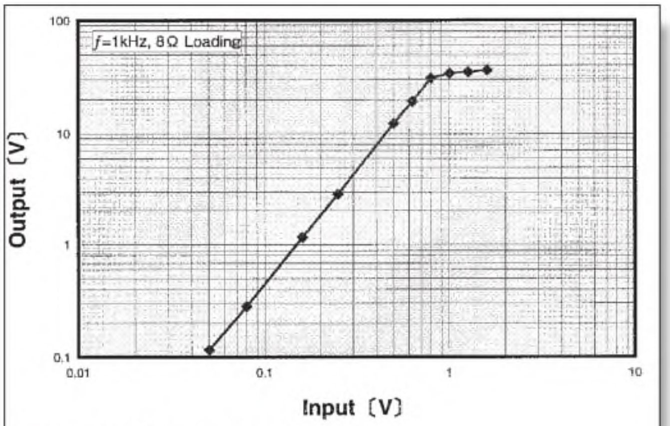

FIGURE 12: Input versus output.

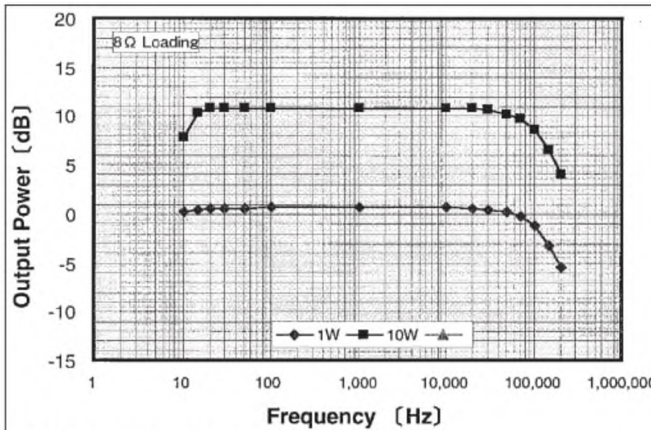

FIGURE 14: Frequency response.

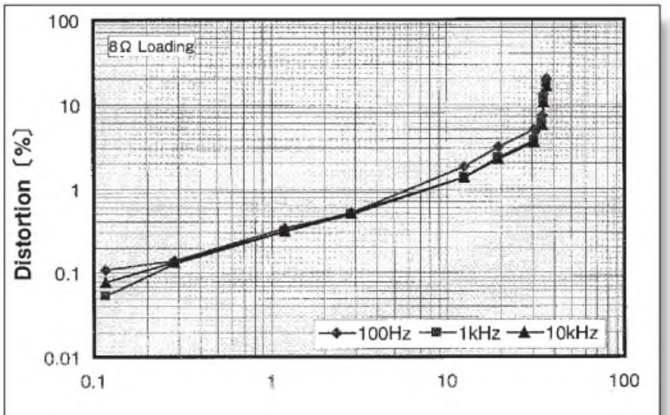

FIGURE 13: Distortion.

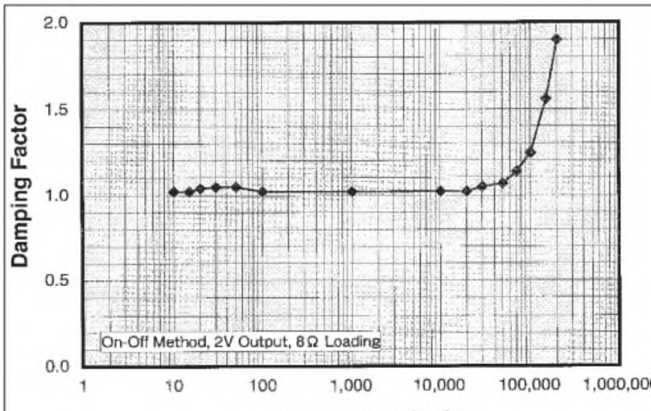

FIGURE 15: Damping factor.

protect my speaker system from possible damage from the buzzing, I disconnected it from the amplifier, leaving the amp to warm up for ten minutes or so. When I reconnected my speakers to the amplifier, the buzz had stopped.

Curious as to why this happened, I sent an e-mail to Eric Barbour at Svetlana and received a prompt reply. His explanation was "Each 3CX300A1 tube has its own warm-up time. The value varies individually, so the deviation of warm-up time may cause imbalance of the plate current. This can cause most or all of the hum and noise in the HV DC supply to appear across the primary of the output transformer, so it does not cancel. This may cause the buzz at the secondary." My curiosity was thoroughly cleared up by his reply.

In addition to this evidence, I found another cause for the buzz, which came out of the MOSFET ripple-filter circuit. I tested this circuit with an external Sorensen power supply (600V 0.75A DC), which decreased the buzz level better than the internal MOSFET power supply, and also decreased its duration.

I found that the root cause was the MOSFET ripple-filter circuit. It became "hung up" just after turning on the power. Approximately 250mA flows out through this circuit into the amplifier, i.e., the output voltage stayed at 420V or so. This was caused by the pinch-off effect of the MOSFET, since a 6V zener diode clamped the voltage across its gate and source electrode.

To eliminate this symptom, I replaced the 6V zener with a 15V zener diode. After this modification, the buzz sound decreased to about one-tenth, though it has not cleared up completely.

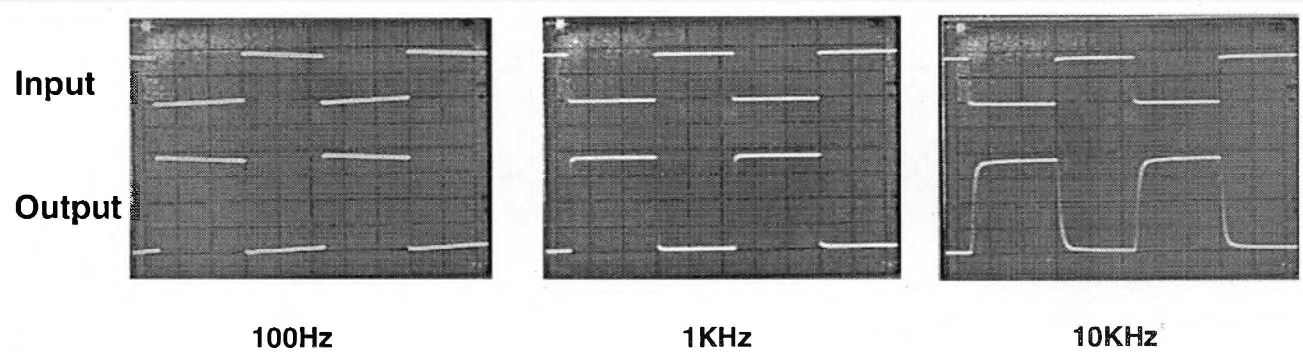

FIGURE 16: Output waveform.

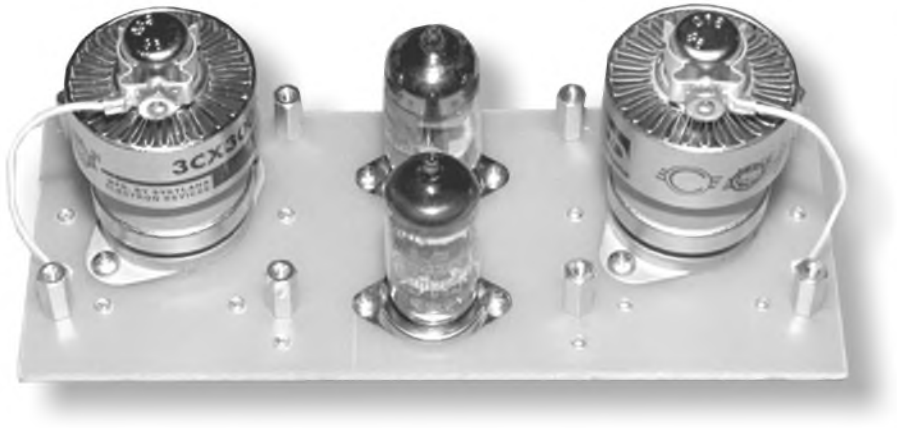

PHOTO 6: All of the tubes neatly sit over the rigid board.

After one amplifier is turned on for 20 seconds or so, this buzzing sound appears, although the other amplifier does not do this. It is tolerable, though I suggest that anyone who hates this buzzing should provide a switching circuit to disconnect the speakers for 30 seconds or so by using a timer relay.

VOLTAGE-DRIVER CHARACTERISTICS—FIGS. 9, 10, 11.

The first stage consists of the differential amplifier, without any AC balancing resistor as is commonly used on long-tail phase-splitter circuits. To verify the well-balanced circuit performance of this driver circuit, I measured the characteristics of both outputs of the differential amplifier. *Figure 9* shows the results.

The difference in output level at the diff-amp outputs is 1% or less, and mostly zero in the range of 0.6V-1V input level, though it will be over 10% when clipping occurs at approximately 1.5V input. In fact, this differential amplifier works as well as any properly balanced phase-splitter circuit, at less than 1% distortion. Furthermore, its frequency bandwidth extends up to 200kHz at –3dB. The gain of this circuit is about 36, and shows a good match with the simulation result.

OVERALL CHARACTERISTICS

1) Input-to-output characteristic—Fig. 12

Clipping occurs at 0.9V input level, giving a maximum power of 35 to 36W. This does not fit with the calculation result of 48W, but it is close to the initial target.

2) Distortion characteristic—Fig. 13

Measured distortion values are 0.3% at 1W, 1.3-1.5% at 10W, and 3-5% at 30W. Overall, this is very good performance for a non-NFB amplifier.

3) Frequency response—Fig. 14

The measured frequency bandwidth was beyond 100kHz at 1W output, and even at 10W. This characteristic depends mainly upon the frequency response of the driver stage. According to van der Veen, the Plitron output transformer is capable of a more extended frequency response than this.

4) Damping factor—Fig. 15

The damping factor is relatively low: about 1, through the entire frequency range. In spite of this, the sound is very powerful at low bass, and highs are clear and transparent. This is due to the distinctive characteristic of the Plitron transformer, mainly its excellent damping at frequencies over 100kHz.

5) Output waveforms—Fig. 16

The waveforms are very distinct from those obtained with a conventional E-I cored transformer. This Plitron toroid does not show any overshoot of the square wave at all. Such performance is similar to that of rare and costly old transformers, such as the legendary Acrosound TO series of the 1950s.

As a miscellaneous fact, the output noise and hum levels of the two mono amplifiers, with no input signal, were 0.5mV and 0.6mV, respectively—a very low noise floor.

LISTENING IMPRESSIONS

At last, you reach the final stage of the assembly, after fine-tuning the power supply. During the final listening tests, I used a homebrew switch box in alternating two amplifier outputs to compare the sound against the reference, my homebrew 300B single-ended amplifier, which also uses a Plitron output transformer.

The first impression of the sound is the strong presence in the vocals by female singers such as Rita Coolidge, Natalie Cole, and the like. Compared to a 300B amplifier, I believe the sound of this circuit is stronger, with crisper detail.

According to van der Veen, the sonic difference is caused by the profile of the harmonic distortion (consisting of the even harmonics produced by a single-ended amplifier versus odd harmonics made by a push-pull amplifier), as well as by the differences in damping factor.

This amplifier shows a very low damping factor—which is less than one due to the non-NFB circuit. Yet the sound of this amp is very strong and vivid. Overall, I was impressed that this newly introduced Svetlana ceramic tube works so well with this stock Plitron toroidal transformer, and generates such a crisp and powerful sound. ❖

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3. Kevin O'Connor, *Principles of Power*, Power Press Publishing.
4. John D. Ryder, *Electronic Engineering Principles*, 1952.
5. Tube CAD manual, Glassware.

SOURCES

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E-mail: smlparts@smallparts.com
http://www.smallparts.com/
Teflon sheet

A Phono Pre-Preamplifier for the CD Era
Part 1

This small phono preamp design by a noted author solves several problems inherent in today's audio systems. **Norman E. Thagard**

While I have designed and constructed several audio power amplifiers in the past 12 years, I have never, until recently, done much preamplifier work. I have considered attempting to design and construct a preamplifier for several years, but I find I am usually prompted to begin a project by need, rather than intent. That is true of the phonograph preamplifier I describe here.

THE NEED ARISES

Unlike some audio purists, I have been very interested in surround sound since the early '70s. With the incorporation of my sound system into an audio/video system, I replaced my Krell preamplifier first by an Adcom and then by a Marantz A/V tuner/preamplifier. Unfortunately, as with most modern units, neither the Adcom nor the Marantz had a phono input. Lately, the Krell has functioned solely as a pre-preamplifier, that is, as an interface between the phono cartridge and a high-level (Tape 1) input on the Adcom.

Having a bulky preamplifier in the system was undesirable, however, because I had no more room for additional components in the equipment rack. I wished to replace the Krell with a dedicated small phono preamp that I could locate behind the equipment rack.

It occurred to me that I was probably not the only one who might need a standalone phono preamplifier that would allow updating the system preamp without losing the phono option. Thus I finally developed the impetus to attempt the design of at least one component of a preamp, the phono stage.

The design I present here allows the phono cartridge to appear as a high-level source that you can input into any of the standard, flat-response, high-level inputs on an audio or audio/video preamp or receiver. The few purists left will appreciate the irony of using the CD input, as I do with the Marantz. The CD input circuitry in my 1985-vintage Krell PAM-5 manipulates the signal in some way—perhaps to compensate for the interchannel time difference in early CD players—so a tape or auxiliary input might be better in some cases.

I specifically designed this preamp for moving-magnet cartridges capable of at least a couple of millivolts output at 1kHz. With some minor changes, you could probably use it with a moving-coil cartridge whose output was 20dB less. I make no claims for this, however, because I did not examine the possibility.

EQUALIZATION

The principles of preamplifier design do not differ greatly from those of amplifier design, the major differences being power and noise considerations. The exceptions are for preamps intended for tape heads and phono cartridges. The cartridge requires not only more amplification (because its output is significantly lower than, say, a CD player), but also equalization.

Equalization involves tailoring the frequency response of the preamplifier in a manner that exactly compensates for the frequency response of the signal at the input. The idea is that the overall frequency response will be "flat" in the sense that the original spectrum of the sound is maintained in reproduction.

The frequency shaping that must take place in the phono preamp is that of the RIAA curve. This curve has been the basis of several previous articles in *Audio Amateur*, so I do not intend to be overly repetitious. I have liberally footnoted this article with references that contain additional information for the reader who seeks more details.

I shall be slightly repetitious in reminding you that the equalization is required because neither the recording nor the cartridge response is flat. To avoid overly wide groove excursions on the record, the low-frequency components are compressed, while, to improve signal-to-noise ratio, the high frequencies are emphasized.[1]

On the reproduction side, a magnetic cartridge is a velocity-responding device. If groove amplitude were held constant, electrical output would be proportional to frequency. To compensate for these factors, response is boosted by a factor of about ten (19.3dB) at the low end and attenuated by roughly the same factor at the high end (19.6dB), where 1kHz is the reference (0dB) frequency.

CURVE-SHAPING POLES

Suffice it to say that there are two poles, one at 50Hz and the other at 2k122Hz, as well as a zero at 500Hz, that shape the curve. There are many different ways to produce these poles and the zero, with different designers favoring different schemes. The most economical method uses negative feedback around only one gain block, with the feedback network made frequency-dependent by adding two capacitors and one resistor to the usual passive voltage-divider feedback network. You can easily accomplish single-ended designs with just two bipolar junction transistors (BJTs) per channel.

Since closed-loop gain is approximately equal to $1/\beta$, where β is the fraction of the output fed back to the input, it follows that a zero in β (i.e., a feedback zero) produces a closed-loop-gain pole, and a feedback pole produces a closed-loop-gain zero. Thus, there are two feedback zeroes

About the Author

Five-time astronaut Norman Thagard was the first American to enter space aboard a Russian rocket for a 90-day mission to the space station Mir. With a total of 140 days in space, he became the most experienced US astronaut ever. In addition to an MS degree in engineering science from Florida State University, he holds a doctorate in medicine from the University of Texas Southwestern Medical School. Dr. Thagard is currently Professor and Director of College Relations at the FAMU-FSU College of Engineering. An avid audiophile, he designs and builds audio amplifiers as a hobby.

and a feedback pole at the frequencies corresponding to the RIAA poles and zero, respectively.

Achieving great accuracy in RIAA curve tracking with this method is more problematical, because the components in the complex RC feedback network interact.[2] It is not a simple case of calculating the three breakpoints as independent RC time constants. Mathematically, there are cross-product terms in addition to the terms of interest. Then, too, if the amplifier is noninverting, the closed-loop-gain rolloff due to the feedback zero at 2k122Hz cannot be sustained, because this gain cannot drop below unity. In effect, an unintended zero is added to the response at some higher frequency.

While you can eliminate this latter defect with an inverting configuration, the ubiquitous 47kΩ moving-magnet loading resistor winds up in series with the cartridge's own DC resistance instead of in parallel with it. The upshot of this is that the considerable noise generated by this resistor is input into the preamp rather than being largely shunted to ground through the source resistance of the cartridge, as it is in the noninverting configuration.

DESIGN PHILOSOPHY

You can find a discussion of the topics of the last paragraph and the philosophy upon which the design described in this article is based in National Semiconductor Applications Note AN 346.[3] This design largely obviates the problems to which I've alluded. Previous articles in *TAA* or *AE* have offered designs based on this same philosophy. Since that is the case, it is valid to ask why I offer yet another similar design. The answer is that I desired to implement a discrete version of the op-amp-based design from AN 346.

It is a good challenge because the AN 346 design used two op amps. Seeking op-amp performance with discrete components can involve a lot of parts, and implementing two such amplifiers in discrete form means yet another such group of parts. Near-op-amp parameters are, in fact, required because high input impedance, low output impedance, and high open-loop gain are crucial to proper performance of the circuit.

As an engineering professor, I receive textbooks for evaluation from time to time. One of these described a one-stage op amp,[4] realized by using the folded-cascode topology. It could be argued that a cascode, folded or telescopic, is really two stages. However, from the standpoint of audio-frequency performance, a cascode has all the earmarks of a single, albeit compound, stage. It behaves like a "super" common emitter or common-source stage with higher bandwidth and lower distortion.

I have had a predilection for cascoded stages ever since reading an article on the subject by Nelson Pass.[5] My very first amplifier design was a 100W Class A (at 8Ω) DC monoblock that featured four telescopically cascoded stages, including the power output stage.[6]

In 1992 I designed and constructed a balanced-input amplifier that featured a MOSFET-based differential-in, single-ended-out folded cascode very similar to the phono preamplifier described here. That design permitted a CMRR >60dB at 20kHz and will be the subject of a future *AE* article. Finally, the Thagard/Pass A75, the design of which was completely Nelson's, but which incorporated some topological features that I had suggested, offered the option of a folded cascode.[7]

COMPLEMENTARY SYMMETRY

In the current application, using a folded cascode in complementary symmetry results in sufficiently high open-loop gain, since the drain of each common-gate device "sees" the drain of its complementary partner as its load device. From the diff-amp input to the single-ended output, gain is given by $\frac{1}{2}g_m R_{load}$, where g_m is transconductance and R_{load} is the load resistance.

If R_{load} is the intrinsic resistance (call it R_o) looking into a JFET drain, then the gain is very high, because R_o often ranges from 100kΩ to 1MΩ. Of course, the caveat is that the input of the feedback network does not load the output node, and that is not a good assumption here. Similarly, the following stage must not load the output node, either, and that *is* a reasonable assumption here. Apparently, the JFET transconductances are high enough to ensure that the open-loop gain is also sufficiently high.

I began with the intention of using the LM394 BJT "superbeta" monolithic matched pair in the diff amp. JFETs typically have a transconductance that is an order of magnitude smaller than that of a BJT in a given application. Also, the source resistance of a moving-magnet phono cartridge is seldom more than about 1kΩ, so that a low-noise BJT such as the LM394 actually has an advantage from the standpoint of noise.[8]

The lower transconductance of most small-signal JFETs significantly adds to their noise voltage output.[9] The JFETs used in this design have values of g_m closer to those seen in similar BJTs, so the LM394 noise advantage will be less. Another entry on the negative side is that the input-voltage offsets and drifts are higher in JFETs. The interelectrode capacitances are higher in JFETs, too, but the cascode largely obviates this negative aspect.

It was Erno Borbely's article describing the 2SK389/2SJ109 high-transconductance complementary monolithic pair JFETs that changed my mind.[10] For one thing, I was aware of no pnp equivalent to the LM394. For another, the BJT input resistance is much smaller than that of the JFET, although the LM394's high beta does result in a respectable resistance.

Thus, this design is mostly a JFET one. It is entirely JFET-based, if you accept the notion that the BJT active-load devices aren't really in the signal path. I believe that this is valid, since, ideally, no signal currents could pass through these devices.

NOISE

I have never given much thought to noise in my power amplifier designs. Indeed, there was little reason to do so for a gain-of-twenty amplifier that requires an input of around 1V in order to produce full-power output. This is why one of my designs uses MOSFETs in the input diff amp. No one can make a good case for low-noise operation of the MOSFETs that would be used in a power amplifier.[11]

Low-noise operation is a major consideration in designing a preamplifier for moving-magnet cartridges. Gain at 1kHz is around 38dB, and the input signal might be about 5mV at best, depending on the cartridge. The output will be boosted by another 20dB in the line-level preamplifier stage and a final 26dB by the power amplifier. This is a total amplification of almost 16,000. Signal or not, any noise at the input is amplified by this factor, which is why even my Krell has some audible noise in the phono position if your ear is close enough to the loudspeaker.

It should be obvious that the first stage in this chain is the most critical from the standpoint of noise. Paralleling devices reduces noise, so the complementary nature of the circuit helps, since the p- and n-channel devices are in parallel as far as the signal is concerned.[12] Unfortunately, a diff-amp stage has 3dB more voltage noise than a single-ended stage, and the two effects cancel.[13]

It may come as a surprise to know that the biggest contributor to noise in a well-designed phono preamp is the Johnson noise generated by the DC resistance of the cartridge itself. For the V15 Type V

MR that I use, this resistance is 1kΩ, for a contribution of about 4nV/√Hz. According to the PSpice model of the circuit of this article, total input noise (all noise is, by convention, referred to the input) is little more than 5nV/√Hz at most frequencies in the audio band.

If the model is correct, the preamp itself adds little to the total noise output. Although at the time of this writing I have made no noise measurements, listening indicates that noise is inaudible unless the ear is positioned immediately in front of the high-frequency drivers of the loudspeakers.

RADIO-FREQUENCY INTERFERENCE

Figure 1 is the schematic of the preamp. Starting naturally at the input, a series inductor stands out prominently. You just do not encounter inductors all that much in audio-frequency design. I had no intention at the outset to use this inductor. However, while doing listening tests during the breadboard stage, I heard not only the record that was playing, but a local radio station as well. I remembered a reference to this possibility.[14] I used the solution suggested in the reference, namely the addition of a 10mH rf choke.

A side effect of using the choke is a slight rise in high-frequency response that causes an RIAA tracking error of about 0.1dB at 20kHz. This rise is due to the interaction between the input capacitance and the inductor. It is possible, of course, to lower the 2k122Hz pole to increase the 20kHz attenuation and thereby eliminate the induced error at 20kHz. I doubt that it is worth the effort.

You can omit the choke if you wish. For compactness, I ordered a choke with a DC resistance of nearly 100Ω. This resistance is in series with the DC resistance of the phono cartridge and will add to the input noise. There are coils available with much smaller resistances—for example, the M5942 from Digi-Key at 7.30Ω. This coil is significantly larger, though, and will be difficult to accommodate if you use the PC-board pattern of *Fig. 2*.

Since a designer need design only to the interface conditions specified by the cartridge manufacturer, I have not been motivated to research the subject, but I have the impression from articles and from cartridge manufacturer-recommended capacitance values that the frequency response of a moving-magnet cartridge is, by design, extended by the resonance of the cartridge's intrinsic inductance with any capacitance in the preamplifier input circuit. If so, this is similar to the "peaking" that is sometimes used in rf circuits to postpone high-frequency rolloff.[15]

The inductance of my cartridge is 425mH, so if peaking is already intentionally used, adding approximately 2% to the cartridge's own inductance may well be less than the variation in intrinsic inductance from cartridge to cartridge. It would not then be reasonable to seek such accuracy in an area where other uncertainties exceed the likely gain.

Since the recommended load conditions for the V15 Type V MR are 47kΩ in parallel with 250pF, I added 100pF in the form of C1. That is because I use an SME 3009 Series II tonearm with 135pF of cable capacitance, and the preamp has about 80pF of intrinsic input capacitance by measurement, or 40pF according to the PSpice version of the circuit.

Even though 100pF is a bit higher than the calculated optimum, I thought that this was probably a good all-around value. You are free to alter the value of C1 for the particular cartridge at hand. Frequency response suffers, however, if there is too much deviation from the recommendation for the particular cartridge used.

TOPOLOGY

The input stage is a dual or complementary differential amplifier. I had hoped that the diff amp input would provide sufficient thermal stability to allow a true DC design. However, to find the zero tempco of the JFETs requires a knowledge of their pinch-off voltages.[16] Since this parameter varies from transistor to transistor, it isn't practical to base this design on the zero tempco. Although I was able to unbalance the diff amp to achieve zero-output offset voltage, the drift was excessive.

In the end, I adopted the same DC servo used by many others before me.[17] Even a venerable old μA741 op amp in the servo gave an acceptably low DC-output offset and drift. However, I accede to the almost universal use of the LF411 in

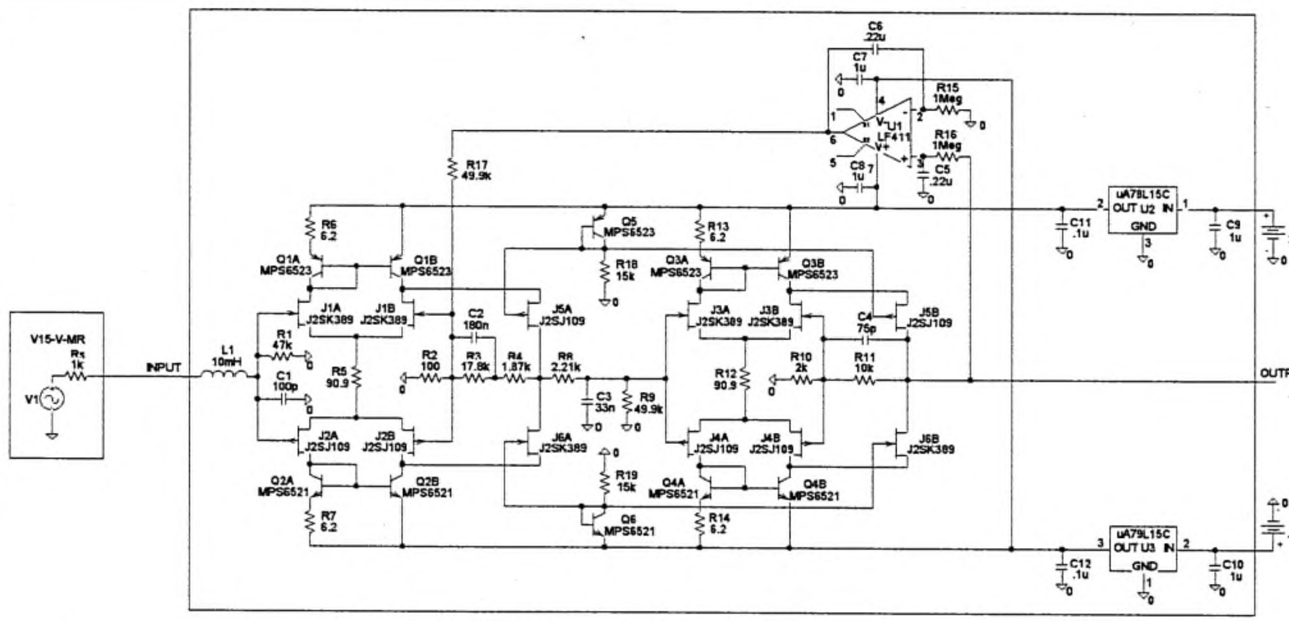

FIGURE 1: Preamp schematic.

specifying that part for this application. Certainly, the LF411 is a good choice, since it is sold as a low-offset, low-drift component.

On the breadboard, the LF411 required bypass capacitors near its power-supply pins, even though the supply rails were already bypassed at the point of entry onto the board. That is why there are two bypass capacitors on the supply rails, one at the output of each voltage regulator and another near pins 4 and 7 of the LF411. You are welcome to experiment with eliminating the pin-4 and pin-7 capacitors. I included them on the prototype as a conservative measure. The μA741 did not require additional bypassing.

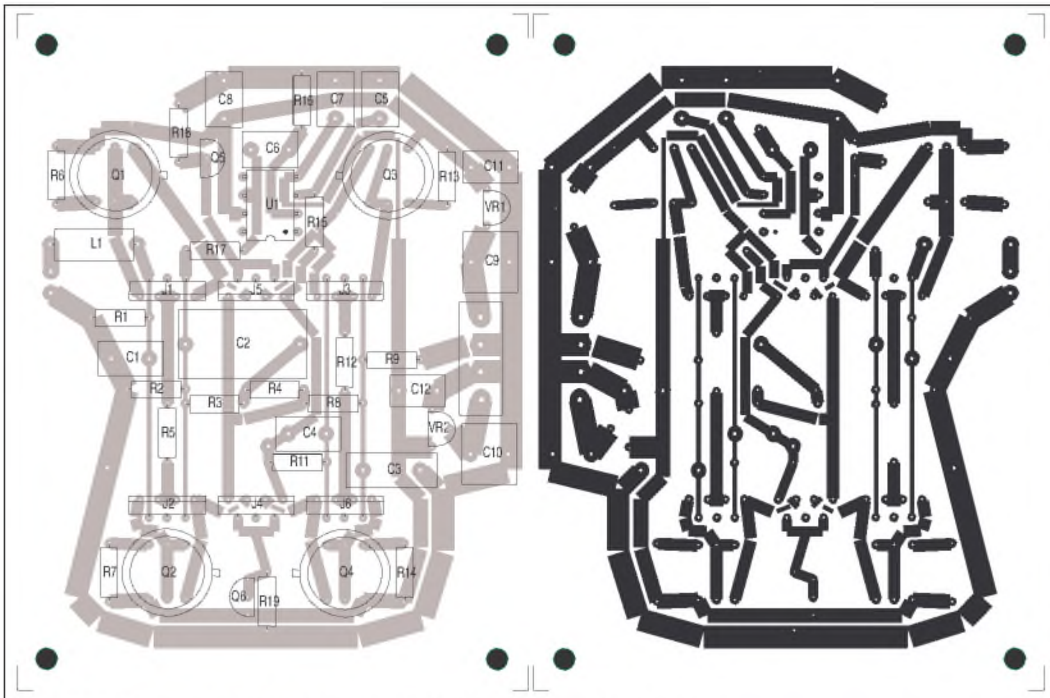

FIGURE 2: Preamp board parts placement (left) and PCB pattern (right).

Be aware that the PSpice model showed considerable low-frequency distortion because of the servo. Increasing the integrator and low-pass-filter time constants corrected this problem in the model. Since I saw no such distortion in the breadboarded circuit, I left the components at the same 1MΩ/220nF values that I have seen used in other designs. However, feel free to increase the time constant by increasing the value of C to 1.0 or even 2200nF. PSpice permits perfect device matching, so the simulated DC-coupled, servoless circuit had almost no DC-output offset or drift, facilitating the determination that the apparent problem was with the servo.

JFET SELF-BIASING

As with vacuum tubes, you can use self-biasing with JFETs. That is the function of resistors R5 and R12. The target drain current was 3mA, meaning that 6mA should flow through those resistors. At 3mA, the JFET characteristic curves I observed on a curve tracer suggested that $|V_{GS}| \cong 0.27V$ was required. With R5 = R12 = 90.9Ω, about 0.55V source-to-source will be dropped.

Since the DC gate voltages are fixed at ground potential, the desired V_{GS} for both n- and p-channel devices is attained, and, consequently, the desired drain currents are realized. JFET characteristics can vary, so if you determine that the drain currents differ greatly from the design value, simply change the value of R5 and/or R12 until $V_{R5}/R5 \cong V_{R12}/R12 \cong$ 6mA. Of course, increasing these resistor values decreases drain current.

The right-hand diff-amp devices J1B, J2B, J3B, and J4B are the common-source halves of a cascode. The more traditional cascode, originally constructed from vacuum tubes, is now sometimes called a telescopic cascode to distinguish it from the folded cascode. The latter "folds" the cascode over toward the opposite voltage rail by using a device for the common-gate half of the compound stage that is complementary to the common-source device.

There were no complementary vacuum tubes, so it took the advent of the transistor before a folded cascode could be physically realized. It has the advantage of allowing the cascode output to be at 0V DC with proper biasing. It facilitates the construction of single-stage op amps and, in discrete form, is the basis for the current design.

The folded cascode allows some flexibility in setting bias current in the common-gate device. This flexibility does not exist in the telescopic cascode, since the drain current is necessarily the same in both halves of the stage. There are reasons for setting folded-cascode common-gate device bias at various levels depending upon the design considerations.[18] For simplicity and noise considerations, I elected to bias the common-gate devices at the same DC drain current as the common-source devices.

Each of the four dual BJTs, Q1–Q4, is configured as a kind of "reverse" Widlar current mirror in order to establish the bias currents for the common-gate devices in the cascode. Again, it should not matter that BJTs are used in this capacity rather than JFETs, since the whole idea of a current mirror is to pass as little signal current as possible.

Some shun the use of active loads in preamplifier and amplifier designs even for folded-cascode biasing. The reason, at least in some cases, seems to be a claimed adverse sonic effect. I am skeptical that any such effect exists except as a perceptual alteration induced by knowledge of the material to which you are listening. I prefer to use active loads because they ensure balanced diff-amp operation and, consequently, potentially lower distortion than you could obtain with simple resistor loads.[19]

COLLECTOR CURRENT

In discrete design, it is easier to establish a given bias current with a BJT-based Widlar mirror than with a scheme involving JFETs or MOSFETs. In BJTs, it is really base-emitter voltage that sets collector current. That is why the current ratio between the transistors in each half of each of the four mirrors is established by the difference in base-emitter voltages of those two transistors. This difference is $\Delta V_{be} = I_e \times R_e$, where I_e and R_e are the emitter current and emitter resistor, respectively, of the left-hand BJT of each mirror.

A 2:1 ratio is required if the common-gate JFET is to have the same DC drain current as the common-source JFET. That should be evident, since the symmetry of the two halves of each of the diff amps guarantees that their drain currents will be nearly equal at 3mA. Thus, if the collector current in the BJT on the right side of the four current mirrors is greater than that on the left side, the extra current must flow into the common-gate device.

If the right-side current is exactly 6mA, then 3mA will flow into the drain of the diff-amp JFET, while the remaining 3mA will flow into the source of the common-gate half of the cascode. From the Ebers-Moll equation, it can be shown that $\Delta V_{be} = .026 1n(I_1/I_2)$ is the difference required to establish the I_1/I_2 ratio. If a 2:1 ratio is desired, as is the case here, then $\Delta V_{be} \cong$ 18mV will do the trick.

Another good rule-of-thumb to commit to memory is that $\Delta V_{be} \cong$ 60mV is good for a 10:1 ratio. I call these mirrors "reverse" Widlar mirrors because the more usual configuration is to have the reference current higher than the programmed current.[20] Here, the left-hand BJT carries the smaller reference current, and the right-hand BJT serves as the current source by "mirroring" a multiple of the reference current.

Since I_e and ΔV_{be} are known, Ohm's law dictates that $R_e = \Delta V_{be}/I_e = .018/.003 = 6\Omega$. I_e is fixed at 3mA, so increasing R_e will increase ΔV_{be} and, in turn, the current ratio, and therefore the current through the common-gate device. Although 6Ω resistors do exist, they are less common than 5.6Ω or 6.2Ω components, and the latter is closer to the desired value. However, 6.2Ω resistors are not as common as 5.6Ω or 6.8Ω resistors. You may use any of these three values, since it isn't as critical as matching the value, whatever it is.

Unfortunately, I may have had the last of the available precision 6.2Ω metal-film resistors in my own parts bin. Currently available units have 5% tolerance. It is, of course, entirely acceptable to parallel two 12.4Ω resistors, which are available in 1% tolerance. I actually used 6.2Ω resistors in one channel and 6.8Ω in the other with no objective or subjective differences discerned between the two.

GATE-TO-SOURCE VOLTAGE

The common-gate JFETs will assume the gate-to-source voltage that corresponds to the drain current according to their square-law equation. For the complementary devices, this is the same $|V_{GS}| \cong$ 0.27V that was seen for the diff-amp JFETs, because the drain currents are the same. The JFET self-biasing feature almost allows J5 and J6 to assume this gate-to-source voltage simply by tying their gates to the appropriate supply rail. Unfortunately, this places the right-hand current mirror BJTs very close to, if not into, saturation. The upshot is that this will work for some BJTs but not for others.

For instance, all of the 2N3811 and 2N2920 matched dual BJTs that I tried worked with this scheme, while none of the devices specified in the parts list (*Table 1*) did so. Since the 2N3811 is no longer available and the 2N2920 is hard to find and expensive, it was not prudent to specify them as the components of choice. Besides, the spec sheets on even these devices indicated that not every example could be counted upon to work correctly in this application.

To ensure that Q1B, Q2B, Q3B, and Q4B would be in their active regions, I added diode-connected BJTs Q5 and Q6, which, themselves, must be in their active regions. This is because V_{cb} = 0 for these two transistors, whereas saturation is the condition in which the collector-base junction is forward-biased. Thus, the magnitude of the collector-to-emitter voltage for the four current-mirror BJTs involved will be $|V_{GS}| + |V_{be}| \cong 1V$. This is more than adequate to ensure that they are not in saturation.

I should comment that, although the vendor of the BJTs specified in the parts list is an advertiser in *Audio Electronics*, I was informed that their usual requirement is a minimum $100 order per part type. Despite this, they did sell me a $50 quantity of each of three different transistors.

SUBSTITUTING DEVICES

Since the question of substitutions will invariably arise, I shall state in advance that, while low-noise matched dual BJTs are preferable, you can use matched low-noise discrete units. NTE, RCA, and other replacement manufacturers sell matched pnp and npn complementary devices. Even unmatched devices may work, especially with the servo in place.

There is sufficient flexibility so that somewhere, somehow, you may find appropriate substitutions. It may also be difficult to obtain the JFETs from US suppliers, although I noticed that Erno Borbely was offering them for sale. Again, being an EE professor has its advantages, since I was able to persuade a wholesaler to sell to me both the n- and the p-channel devices in quantities of 100.

Unfortunately, it was necessary to take what was available, and the n-channel and p-channel devices that I used are from dif-

TABLE 1: PARTS LIST

PREAMPLIFIER (one channel only)	
Resistors (¼W, 1% metal film unless otherwise specified)	
R1	47k0Ω
R2	100Ω
R3	17k8Ω
R4	1k87Ω
R5, R12	90.9Ω
R6, R7,	6.2Ω
R13, R14,R8	2k21Ω
R9, R17	49k9Ω
R10	2kΩ
R11	10kΩ
R15, R16	1MΩ
R18, R19	15kΩ
Capacitors (Panasonic P-Series 50V. Polypropylene unless otherwise specified): Four 0.01mF disc ceramics (two for phono-input bypass, two for AC line filter of power supply).	
C1	100pF, 5%
C2	180nF, 2%
C3	33nF, 2%
C4	75pF, 5% silver mica
The following capacitors are Philips 63V, 5% metalized polyester film:	
C5–C8	220nF
C9, C10	1µF
C11, C12	100nF
INDUCTORS	
L1	10mH RF choke
SEMICONDUCTORS	
J1, J3, J6	Toshiba 2SK389 low-noise n-channel monolithic dual JFETs
J2, J4, J5	Toshiba 2SJ109 low-noise p-channel monolithic dual JFETs
Q1, Q3	Linear Systems LS 352 low-noise pnp monolithic dual BJTs
Q2, Q4	Linear Systems LS 312 low-noise npn monolithic dual BJTs
Q5	Motorola MPS 6523 low-noise pnp BJT
Q6	Motorola MPS 6521 low-noise npn BJT
INTEGRATED CIRCUITS	
U1	LF 411 low-noise, low-drift op amp
U2	µA78L15 +15V, 100mA voltage regulator
U3	µA79L15 –15V, 100mA voltage regulator
POWER SUPPLY	
Resistors	
R101, R102	220Ω, 2W, 2% metal
Capacitors	
C101, C102	2,200µF, 50V aluminum electrolytic
C103, C104	4,700µF, 35V aluminum electrolytic
Transformer	
T1	48Vct at 150mA (see text)
Fuse	
F1	0.05A (see text)

ferent IDSS groups. I was able to find a sufficient number of complementary devices, nonetheless, through the use of a curve tracer. If possible, choose all "V" or all "BL." I do not recommend transistors from the "GR" group because of the possibility that some units have values of IDSS that are too low for this application.

You should certainly avoid getting one type from the "GR" and the other from the "V" group, since there is no IDSS overlap between the two. In quantities of 100 each, the prices drop to less than $1 per transistor. I am not at all sure that there are good substitution alternatives to these JFETs. Perhaps *audioXpress* will make some of these devices available if there is sufficient interest.

FEEDBACK-LOOP GAIN AND FREQUENCY RESPONSE

Notice that the output of the first-stage cascode is fed back to the input. No second or third stage is enclosed in the feedback loop. Also note that the feedback network is not frequency independent. A 180nF capacitor, C2, is added. At low frequency, this capacitor is effectively an open circuit, and the gain is

$$A_V = \frac{R2 + R3 + R4}{R2} = 1 + \frac{R3 + R4}{R2} = 198\,(46\text{dB}).$$

As frequency increases, the capacitive reactance X_{C2} will decrease until it equals R3. This defines the 50Hz breakpoint of the RIAA curve. This breakpoint is a zero in the feedback factor because the effect of decreasing capacitive reactance shunting R3 is to increase the amount of feedback. As frequency increases still further, it finally reaches a point where an additional decrease in the capacitive reactance causes very little change in the feedback factor.

When the feedback factor is within 3dB of its ultimate and maximum possible value, the second breakpoint is reached. This is a function of all three resistors and the capacitor, and occurs at 500Hz with the component values used.[21] You will recognize this as the 500Hz zero of the RIAA curve. It is, of course, realized as a pole in the feedback factor, since it represents the point at which the rising feedback caused by the feedback zero at 50Hz is cancelled.

In the absence of other higher poles and/or zeroes, the closed-loop response above 500Hz would be flat. This approach, then, avoids the unintended high-frequency zero of the noninverting single-stage RIAA amplifier that was mentioned at the outset.

EVALUATION FROM A FRIEND OF THE AUTHOR:

I got your preamp hooked up this evening into the Tape-2 input. It sounds great—very quiet and no hum at listening levels. The gain is definitely less than my PS Audio phono stage: I'm running the volume control at the 11:00 position, whereas it's normally at 8:00 for phono listening. I tried to match listening levels between your phono stage and my normal phono setup for a fair A/B comparison. Using Paul Simon's Graceland album, I noticed a wider and more three-dimensional soundstage, which seemed to be placed a bit lower. I don't recall any significant differences in musical detail, and it was definitely a clear, transparent sound. —Mitch

An important design consideration is the absolute value of resistor R2. After it is set, the values of R3, R4, and C2 are set based on the desired closed-loop gain, the pole and zero frequencies, and the value of R2. Like R1, R2 can be a major contributor to circuit noise. Therefore, from a noise reduction perspective, it should be as small as possible.

Unfortunately, you are faced with yet another trade-off, since, ideally, the input resistance of the feedback network would be infinite. However, the lower the value of R2, the lower this input resistance will be, since R3 and R4 must be scaled accordingly. Power amplifiers have no problem driving low-impedance feedback networks, but many, if not most, preamps would.

OPEN-LOOP GAIN

Open-loop gain can also suffer if the feedback network loads the output. Low open-loop gain a or low-feedback fraction β can lead to deviations from the predicted value for closed-loop gain if, as a result, the closed-loop gain formula $A_{fb} = a/(1 + \beta a) \cong 1/\beta$ becomes a poor approximation. $\beta = 0$ for the no-feedback case, so that $A_{fb} \equiv a$.

Open-loop gain is usually quite variable even among apparently identical amplifiers. As frequency decreases in the first stage, closed-loop gain rises, reflecting the decreasing value of β necessary if the rising low-frequency response dictated by the RIAA curve is to be realized. Feedback, which is the difference between the open- and closed-loop gains, is being squeezed. This, combined with the relatively low open-loop gain, means the assumption that $\beta a >> 1$ (which is implicit in the approximate closed-loop-gain equation) must fail.

Precise RIAA tracking at low frequency is predicated upon $A_{fb} = \beta^{-1}(f)$. Since there was almost no error, even at 20Hz, the open-loop gain must still be adequate despite the potential problems just discussed. There is absolutely no reason you could not use the component values given in AN 346, and doing so ameliorates the potential problem discussed in this paragraph. Since performances of two units were identical and precise, I decided to use the 100Ω value for R2, and I proceeded from there.

While I don't intend to repeat the applications note, I do wish to present the formulas for the component values. This will allow you to customize the design, should you desire that. The technique in AN 346 is to set R2 based on the aforementioned trade-offs, choose a 1kHz first-stage gain in the range $10 \le A_{fb} \le 30$ (20 – 30dB), and then, based on these two factors, calculate component values starting with R3 = 8.058R2 × A_{fb}. From there, C2 = 0.00318/R3, and R4 = R3/9 – R2.

The formula derivations, found on the last page of reference 21, follow directly from the transfer function. There is almost always more flexibility in choosing resistor values than capacitor values, so the closest commercial 1% capacitor value may differ enough that it is necessary to recalculate R3 according to R3 = 0.00318/C2.

The applications note then calls for a recalculation of R2. I doubt that is necessary, however, because R2 does not affect the 50Hz RIAA breakpoint at all, and affects the 500Hz zero only slightly. It does have a major effect on the 1kHz gain, but this is not standardized, which is the reason a range is specified in the first place.

PASSIVE EQUALIZATION

There is, of course, another pole you must establish to conform completely to the RIAA curve. It is provided passively by the low-pass filter formed by R8-C3. Actually, the precise calculation of this pole frequency requires the Thévenin-equivalent resistance seen by C3. C3 sees R8‖R9 rather than simply R8, so the technique used in AN 346 is first to choose a 1% capacitor value in the range of 10–50nF, then to compute an R_p = 75µs/C3 = 2k273Ω.

The 75µs in the equation is simply the RC time constant corresponding to a pole at 2.122kHz. Since R8 and R9 are in parallel, choose a slightly larger 1% resistor value for R8. I chose the value of 2k37Ω. Finally, compute R9 so that R8||R9 = R_p. This can be done using R9 = $1/(1/R_p$ –

1/R8) = 55k536Ω. The nearest 1% value is 54k9Ω.

Now I must 'fess up. Resistor R9 was required in AN 346 because the output of the first-stage op amp was capacitively coupled to the input of the second-stage amplifier, which was an LM833 bipolar op amp. Without R9, there is no input bias current path for the LM833. There is no such requirement in this design because the first stage is directly coupled to the second stage.

I left R9 in nonetheless, because I desired a stage that could serve purely as a phono section driving a separate preamplifier line stage through a 50kΩ potentiometer if I should decide to incorporate the section into such a preamplifier in the future. I subsequently realized that if I were to do this, I would still need a buffer.

At this point, I would keep the current topology, but make the second stage a gain-of-ten (20dB) stage and reduce the 1kHz reference gain of the first stage accordingly. This, of course, would require recalculating the feedback-network component values. Lowering first-stage closed-loop gain is another way of easing any problem of low open-loop gain in that stage.

Omitting R9 seemed to make the frequency response—and consequently RIAA tracking—more sensitive to component-value variations. If you nevertheless decide to eliminate R9 altogether, PSpice shows that R8 = 2k15Ω gives the proper attenuation at high frequency. It also, however, shows that R3 should be simultaneously reduced to 16k9Ω; otherwise the boost at low frequency is excessive. Since it is cheaper and easier to use a number of 1% resistors than 1% capacitors, I suggest that you vary R3 if low-frequency response needs trimming.

Eliminating R9 also ameliorates the low-frequency, open-loop gain degradation by increasing the impedance magnitude that the first-stage output must drive. If you need lower output impedance for either the first or the second stage, you could easily add a follower/buffer. Of course, the whole idea was to keep it simple, and that would be a step in the wrong direction.

The early PSpice versions of the circuit used followers at the output of both stages. I eliminated these when PSpice indicated—and breadboarding confirmed—that the buffers were not essential.

You should note that R9 = 2k21Ω, which is less than the calculated value of 2k37Ω. This difference comes about because the output resistance of the first stage is not 0Ω, as it very nearly is for the op amp-based design of AN 346. The output resistance of the first stage must be folded into the R9 value, or the pole frequency will be lower than anticipated.

Despite the shortcoming of my discrete one-stage op amp in the area of output impedance, I was able to achieve almost perfect high-frequency tracking of the RIAA curve, with the response dead on at most frequencies. The maximum errors measured were 0.1dB more attenuation at 20Hz and 0.1dB less attenuation at 20kHz than would be ideal. I have already alluded to the reason for the error seen at 20kHz.

The best feature is the reproducibility of the tracking. Both channels that I constructed tracked identically, and their gains at the 1kHz reference frequency were the same as the standard 1% capacitors and 1% metal-film resistors used at the breadboard stage. In one channel, I had no 1%, 33nF capacitor, so I hand-selected a 10% unit whose measured value was within the 1% tolerance band.

Since Digi-Key is a convenient source of parts for hobbyists using 2%-tolerance, 50V polypropylene capacitors, I ultimate-ly used Digi-Key components in the prototype, and it is those capacitors for which the PC-board pattern is sized. Again, claims are made for sonic effects due to capacitor dielectric material, but I am skeptical of the existence of such effects. Even so, polypropylene capacitors are objectively superior to most others, and they were the only precision capacitors available from this source.

Both channels of the prototype still track the RIAA curve with 0.1dB accuracy, but they do differ from one another by as much as 0.2dB, probably because of the relaxed tolerance on the capacitors. Since there is a price break for quantity purchases, you could order them in lots of ten, using a capacitor meter to find units that fall within 1% of the design value if slight tracking deviation were a concern.

SECOND STAGE

It is possible to realize the entire 38dB of gain in the input stage, although distortion would be greater. To do so, it would almost certainly be necessary to increase the output-node impedance magnitude by increasing R2 and scaling the other equalization-network values accordingly. In that case, or if you incorporated this phono section into a control preamp with a volume control of 50kΩ as previously discussed, the pot could then replace R9, and you could omit the second stage.

The increased distortion would probably be acceptable, but I did not like the idea of an unbuffered R8-C3 filter output driving an unknown downstream device. Therefore, I faithfully followed the scheme of AN 346 with a second folded-cascode single-stage op amp realized in discrete form. In fact, the second stage is identical to the first except for the feedback network.

All the equalization is taken care of in the first stage and the interstage low-pass filter. However, an additional flat gain of 10 to 20dB is needed to bring the 1kHz gain to a level where the output is in line with that from a tuner or CD player. The gain used here differs slightly from that of the AN 346, which is a result of trimming the second-stage gain by ear.

With an LP as the source, I simultaneously fed the same cartridge channel into the phono input of a Dynaco PAT 5 preamp and my preamp. The output of my preamp was fed into the tuner input on the same PAT 5. I varied the gain of the second stage until the two signal paths produced the same speaker volume. Therefore, I can say with certainty that the output level from this phono preamp is in line with at least one commercial preamp.

There is no output level adjustment, but this should not be a problem. The overall gain is within the range normally chosen for interfacing a moving-magnet phono cartridge with a preamplifier line stage. You can increase (or decrease) the value of R11 if you desire more (or less) gain.

Output Impedance

The magnitude of the output impedance at the drains of J5B/J6B is around 60Ω. This reflects the high open-loop impedance at this node that cannot be lowered below this value despite the large amount of negative feedback to be found in a gain-of-6 configuration. Thus, if the preamp must deliver a lot of current, output voltage will fall rapidly. Still, this is not excessive output impedance for most circuits that would accept this output.

An advantage of the folded-cascode op amp is that load capacitance does not contribute to instability; rather, it enhances stability by lowering the dominant pole frequency.[22] I added capacitor C4 for stability of the second stage, since a high-frequency oscillation existed under no-load output conditions. I did so even though the oscillation, as expected, disappeared with the output connected to a preamplifier.

It is poor practice to offer a design that has even a remote chance of oscillating. As was discussed elsewhere in *Audio Electronics*,[23] an excellent method of compensation, if it works, is to roll off the closed-loop response with a feedback zero such as that produced by the addition of C4. As it turns out in this simple design, pole spacing permits successful use of such a scheme. PSpice did not predict this oscillation, which illustrates that the CAD tools are useful, but have their limitations.

I am convinced that running the output into one additional gain-of-10 cascode stage that's otherwise identical to the second stage of this preamp will yield a perfectly acceptable signal that could directly drive a power amplifier. In other words, the additional stage would serve as the line-amplifier stage, and the control preamplifier could be omitted if this were a phono-only system. For this, a 50kΩ dual-potentiometer should be inserted between the second stage of this circuit and the new line-amp stage to serve as a volume control. You could build a complete control preamp around this topology if you so desired. ❖

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THD Is Meaningless, Part 1

Looking at audio amplifiers from an RF designer's perspective, the author argues that THD figures are irrelevant, irrational, and completely spurious. **By Anthony New**

(Reprinted with permission from Electronics World*)*

A recent article in Britain's *Electronics World* by Ian Hickman showed how to measure total harmonic distortion, or THD, down to levels below 0.001%. This achievement is worthy of applause for its technical challenges, yet I cannot help marveling at the enormous waste of effort that has been made over the years on such an irrelevant, irrational, and completely spurious figure as THD.

Irrelevant? Irrational? How so? And how can a figure used so frequently in audio design be spurious? The latter is a very good question, and one which I have not been able to answer.

I think I can explain why the standard definition of THD is completely meaningless as an indication of what it purports to measure and how it is utterly irrelevant to the uses to which it is generally put. However, I have no idea why the many engineers with far greater experience of amplifier design than myself should continue to use the term at all, let alone attach so much importance to it. Yet they do.

So what are my objections to it? The problems fall into several categories.

TOTAL HARMONIC DISTORTION

First, the concept of "total" harmonic distortion is spurious because it sums a great many separate components which are not equal in kind or effect. Anyone who has experimented with waveform generation will appreciate that, for example, 1% second- or third-harmonic distortion on a reasonably pure tone has a quite different sound from 1% seventh or ninth harmonic, and is much less audible (*Fig. 1*). In fact 1% of second- or third-harmonic distortion is not only not unpleasant but is sometimes positively preferred by those who like "valve sound," whereas early transistor amplifiers producing a great deal less than 1% of higher-order harmonics sound pretty awful on any challenging music.

Second, of course, many of these harmonics will be outside the range of human hearing anyway. It is common practice to include distortion figures at frequencies as high as 5 or 10kHz, but of what possible significance are they? As a young man I could (just) hear loud tones as high as 20kHz, and found the common TV line-oscillator whistle at 15.625kHz acutely painful. But I very much doubt whether anyone can hear the third or fifth harmonic of a 10kHz tone—even a loud one—and certainly not one of amplitude below 1% of its fundamental.

I contend, therefore, that the practice of adding all distorting harmonics together to give a sum total, without any weighting factors, is quite arbitrary and not indicative of the audibility of any harmonic

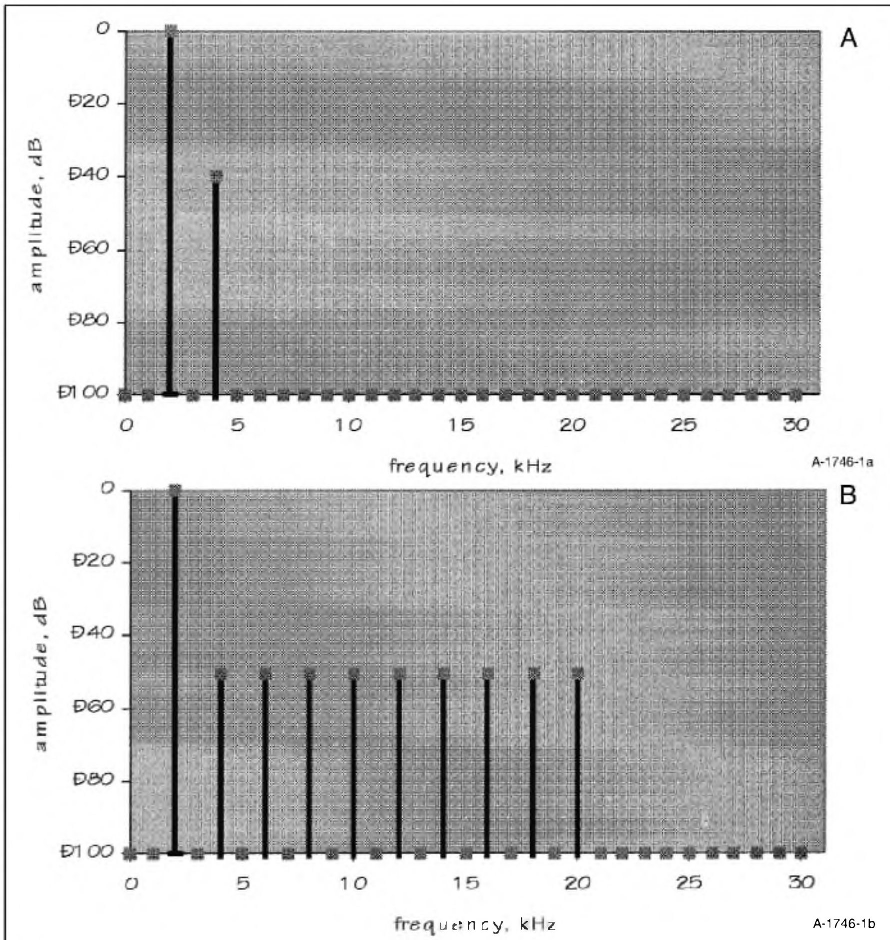

FIGURE 1: Subjective audibility of THD: which sounds worse—1% of purely second-harmonic distortion as in (a) or 0.5% of mixed harmonics as in (b)? Probably the latter, though its THD specification is better. All the spectra that follow have a logarithmic Y-axis (amplitude) and a linear X-axis (frequency) even where this isn't shown.

ABOUT THE AUTHOR

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distortion produced by an amplifier.

Since the audibility of a given THD figure depends heavily on its actual makeup, the figure is also pretty useless even as a purely theoretical comparison of two or more amplifiers, since no acoustic model of audibility is included.

THD AND THE EAR

However, there are yet worse flaws in the THD concept which make the previous problems almost academic. These concern the very nature of harmonic distortion itself.

In my view one of the central problems with traditional audio amplifier design is the insistence on considering the device as a piece of electronic equipment devoid of any psychoacoustic considerations. The extreme of this was the concept of "a straight piece of wire with gain," which is fortunately unattainable, as its gain and bandwidth would make it seriously less than optimal and possibly quite unusable in a real system.

This is not to say I fall into the "subjectivist" camp in audio criticism—far from it. I have listened attentively to the debates those of this persuasion have had with such luminaries as Douglas Self and have been mightily impressed with Self's clear—and seminal—analysis of amplifier distortions.

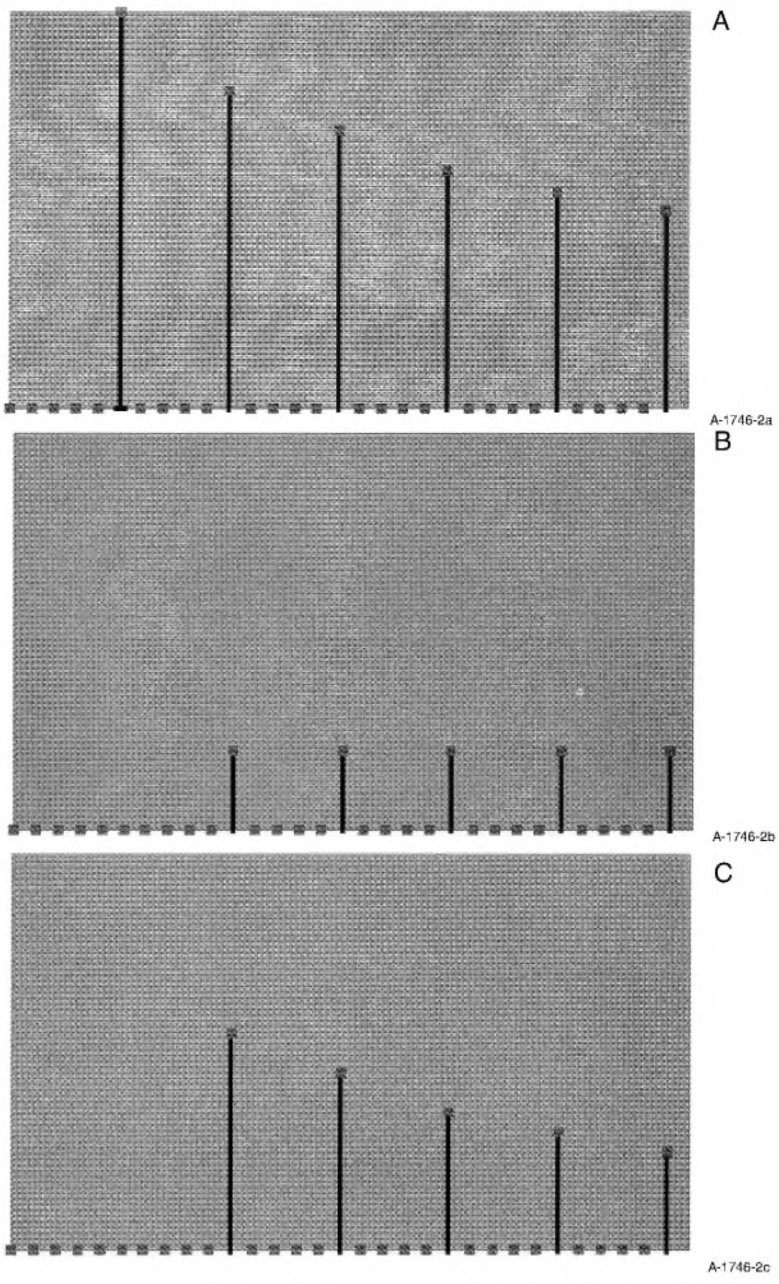


FIGURE 2: Auditory masking of harmonic distortion: (a) typical spectrum of real signal with many harmonics; (b) nominal distortion products at 0.01% each; (c) error products due to frequency-response nonlinearity on original signal; note that these are on the same frequency as the distortion signals and at much higher level, masking the actual distortion.

The problem I have with these debates is that neither side seems particularly interested in what the other is saying. On the one hand, we are told "all the distortions have been correctly analyzed"; on the other "a difference can be heard."

It seems to me that if these opposing views are to be reconciled, the answer must lie at least partly in psychoacoustics, that is—as far as I am concerned here—the study of how we perceive sounds.

I don't claim to have any professional qualifications in this field, but one thing stands out about the current discussion of distortion in audio systems; namely, the lack of any auditory model. It is as if in designing seats and seat belts for cars, nobody was prepared to test a human body—or even a dummy model of one.

I can certainly understand how an engineer is tempted to subtract the input signal to an amplifier from a linear proportion of its output and declare—by definition—any difference to be distortion. The problem I have with this view is that traditional THD testing methods look at only one small part of this difference, and as far as I can see, *harmonic* distortion isn't perceived by the ear as *distortion* at all.

What is the effect to a listener of adding a few percent harmonic distortion to the waveform of a musical instrument or group of instruments? It is to brighten the *timbre* of the instrument.

Since most of the "distortion" products will already be present in the undistorted signal, a similar effect may be obtained by adjusting the tone controls. Those of you who have spent much time siting microphones in the recording industry will be aware how critical their exact placement is to recording balance—and I don't simply mean relative loudness.

You will also be aware of the dramatic change in both *subjective* sound and *objective* frequency response obtained by alterations in these positions. Even small movements can have effects far more noticeable than minute levels of THD in the recording or playback medium.

GOLDEN EARS

It seems quite possible to me that when those with "Golden Ears" say they can hear a difference with such-and-such change in the equipment they may be right.

When I was younger and my ears were sharper, I listened to many excellent loudspeakers. Very few sounded as good as a live performance. I heard only one—namely, the Quad electrostatic—that could actually fool me into thinking the performer was present in the room. The illusion was so strong that I was convinced the performer was hiding behind a curtain until I looked.

Even now in any hi-fi demonstration, the difference in sound between different loudspeakers in the same room—even those produced by the same company—is so marked as to make a nonsense of the claim that many of them can really be "low distortion" in the "blameless" sense that Self used for amplifiers. It also makes a nonsense of the idea that state-of-the art amplifier distortion could be significant compared with it.

The point is that "being able to sense a difference" is *not* equivalent to "sensing distortion" in any meaningful sense. Nor is it an indication even that one of the items being compared is *necessarily* better or worse than another.

Any real musical instrument—including electronic ones such as keyboards—produce sounds which, when converted into analog electrical signals, contain possibly many discrete tones. Usually they also contain many harmonics of the tones, the relative amplitudes of which strongly influence the "sound." The relative levels of these harmonics—both perceived and measured—vary with many factors, including auditorium response and the distance between source and listener. Further factors occur due to the room where the sounds are replayed. Even the shape of the ear itself has an enormous effect, and the presence of hair or hat!

Consequently, even for a particular note played there is no absolutely "right" or "wrong" quantity of any of these harmonics. A *slight* alteration of the levels of these does not correspond to an unpleasant "distortion" of the sound but to a slight change in perceived distance, position, or playing by the instrumentalist.

Furthermore, such slight changes in these levels may be correctable—or at least adjustable, in part—by variation of the user's tone controls. In addition, the recording engineer may already have done this to a considerably greater extent prior to or after mixing the output of several microphones.

FREQUENCY RESPONSE

I also contend that most of the apparent subjective differences that still exist between different audio amplifiers are *not* due to distortion at all but to slight differences in frequency response. This point should receive far more attention during design than it generally does.

Any deviations from a flat response are likely to have a greater impact on the level of high-frequency harmonics present in the amplifier output than the tiny harmonic *distortion* products. If noticeable and uncorrectable with tone controls, these can also contribute to listener fatigue.

The human ear/brain combination is also very good at correlating impressions over time, so even slight bumps in the frequency response can become noticeable and even irritating eventually. Since these real-world variations in the levels of a signal's harmonics dwarf any likely distortion products in a correctly operating amplifier of moderately good quality, it seems perverse in the extreme to use any measure of these tiny "distortions" as a useful figure of merit.

I also note that conventional methods of measuring amplifier performance don't really satisfy the traditional definition of distortion—output relative to input. Distortion tests use only a single frequency source and cannot monitor either nonharmonic distortions or frequency-response errors. Also, the frequency-response tests are

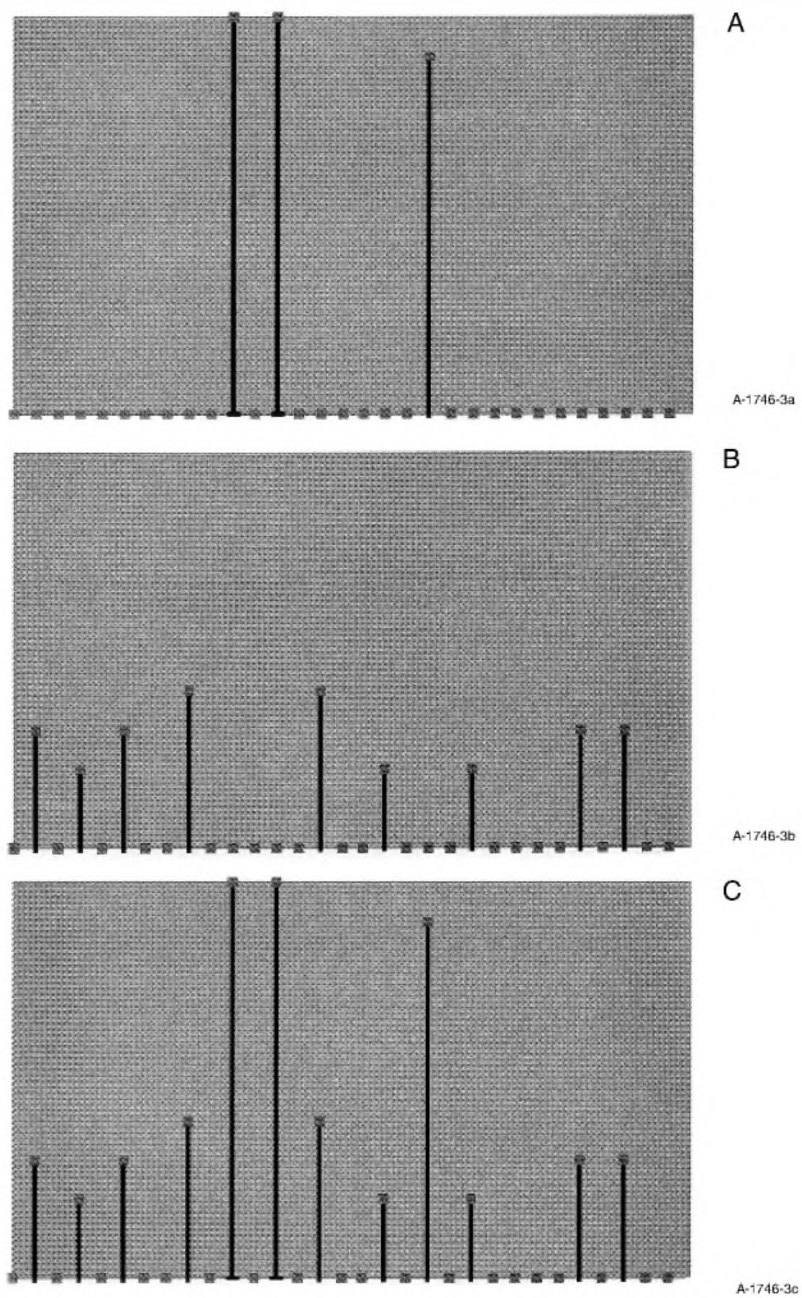


FIGURE 3: Intermodulation distortion showing the lack of auditory masking: (a) pure multi-tone signal at input to amplifier; (b) third-order IMD products produced by amplifier; (c) amplifier output signal. IMD products are at different frequencies from either input tones or harmonics, and therefore not masked.

done differently and are far less sensitive.

For example, when did you last see an amplifier's frequency-response flatness specified to 0.01%? Plus or minus 1dB is more usual, which is 12%, and even 0.5dB is still 6%. Of what possible significance is the 0.001% harmonic distortion of an amplifier when its frequency response contributes an error in harmonic content of several percent?

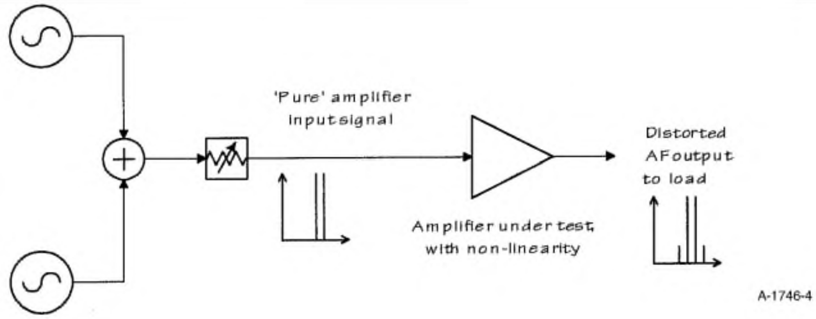


FIGURE 4: Producing multi-tone test signals with standard sinewave signal generators. Harmonic output of the generators is not critical. The passive combiner and attenuator should not affect the measurement linearity—their IP3 can be measured in principle by increasing the generator output level beyond what the amplifier requires, allowing the effective generator IMD to be calculated at the lower levels for the amplifier test.

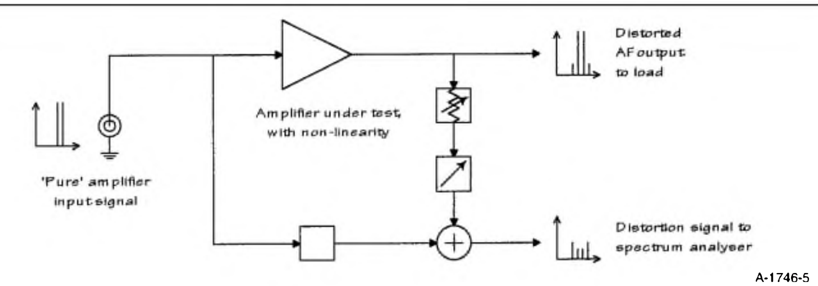


FIGURE 5: Possible distortion measurement circuit. The attenuator, phase shifter, and time delay are first adjusted on a network analyzer to cancel the input signal as well as possible across the whole audio range. This reduces the dynamic range of the distortion signal for spectrum analysis. The residual input tones also reveal the gain flatness of the amplifier over frequency, which contributes to the amplifier's output errors. The input tones may be swept across the frequency range with a constant difference frequency.

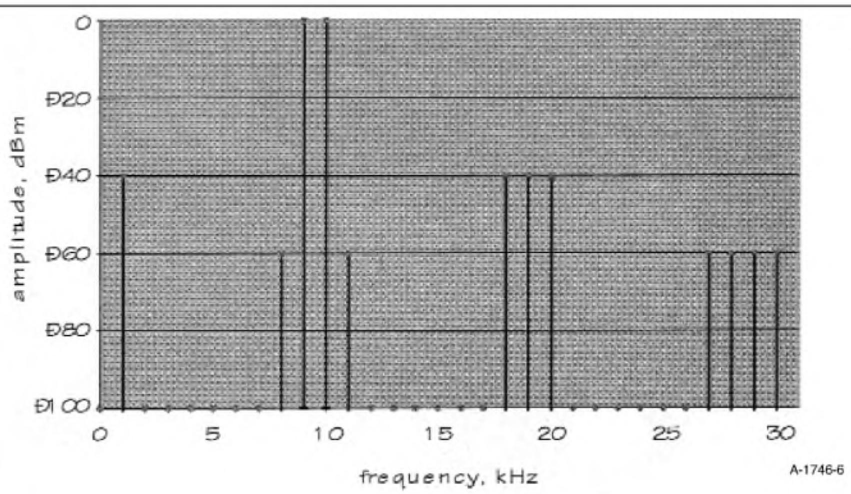


FIGURE 6: Frequency spectrum of two-tone signal showing expected second- and third-order products due to intermodulation distortion. The lowest frequency component is the "beat" frequency between the tones; the two small components next to the two main tones are the "in-band" third-order components; and the rest are a mixture of harmonic and higher-frequency nonharmonic products. In an audio amplifier all these—and more—may be audible for some pairs of tones, though they might be out-of-band in a typical RF amplifier.

Since the harmonic distortion products will also lie on existing signal frequencies, they will be effectively masked from audibility (*Figs. 2* and *3*).

INTERMODULATION DISTORTION

Does this mean that I join the subjectivists in eschewing measurement completely? Not at all. It just means I favor using a *sensible* measure of distortion instead of a *senseless* one. Fortunately, one is conveniently to hand.

Outside of the parochial and fashion-conscious world of audio, most amplifier designers have long since given up measuring or even talking about harmonic distortion and use instead *intermodulation* distortion, or IMD, for short. Measuring IMD has three particular virtues over THD. One is that, unlike THD, IMD is always a measure of distortion in-band. No weighting is needed for audibility at different frequencies.

The second is that it really *does* degrade performance of a system. It does so regardless of whether it is measured *objectively* by such quantities as BER (bit-error rate), SVE (signal-vector error), or spectral spread or regrowth, or *subjectively* by intelligibility of communication.

A third advantage is that unlike the case of harmonic distortion, intermodulation distortion is quite easily measured by standard laboratory equipment (*Figs. 4* and *5*). At a stroke the problem introduced earlier of distortion of 10kHz tones is solved. If two tones at, say, 9kHz and 10kHz are supplied to a good but not perfect amplifier, it is not the *harmonic* distortion that is audible but the *intermodulation* distortion.

The nonlinearity in the amplifier produces new tones, not present in the original, such as, in this case, 1kHz, 8kHz, and 11kHz (*Fig. 6*). Although the audibility of IMD depends on the type of music, in general it is much more audible than any harmonic effects precisely because the distortion produced is not harmonically related to the signals of interest.

Intermodulation distortion typically makes music sound muzzy and indistinct. The worse case of this is usually heard on old car loudspeakers, where the cone is broken or the voice coil rubs on the pole pieces, but it can be heard in very much more expensive and well-cared-for equipment. This is the reason a blameless amplifier must be linear—the harmonic distortion measured is a complete red herring. ❖

Next month the author takes a closer look at IMD measurement as a viable assessment of audio amplifiers.—Eds.

On Angel's Wings, Part 1

Now you can enjoy the best of both worlds by combining the spacious sound of ribbons with the dynamic range of conventional drivers.

By Tom Perazella

You've all probably had the sensation of something visual or audible literally grabbing your attention because the effect it produces in your mind is so striking. I remember that happening to me the first time I heard an electrostatic speaker. Nothing before had ever sounded so utterly detailed or spacious. I was hooked. The sound was so glorious that it blanked out the fact that the frequency response and dynamic range of the sound was limited.

Over time, after listening to a broad range of source material, those shortcomings became all too apparent. My dynamic speakers were capable of reproducing a better physical sense of the performance, although the romance was missing. The desire persisted to have the detailed sound that electrostatics and planer magnetics can produce. I kept asking myself why there were no speakers that could produce this detail while also having the frequency range of classical dynamic speakers.

THE BOHLENDER DRIVER

Enter, stage left, the Bohlender Graebener RD75 driver. It happened while I was in Boston on a business trip. I stopped in to visit a member of the Boston Audio Society with whom I had been discussing subwoofers. As I walked into his listening room, I saw these large, thin drivers standing next to his huge subs. They were mounted in baffles and operated in dipole fashion. Hmmm, I thought. This could be interesting. I asked him what they were, and he proceeded to explain that they were planar magnetics manufactured by a company called Bohlender Graebener (BG).

As soon as the music started, it all became clear. Here were drivers that had the best of both worlds. Detail and spaciousness were wed with dynamic range. And what a marriage it was. I was hooked again. To top it off, I'm a real sucker for dipoles, and here was a pair of dipoles that could rock. I knew that things were about to change for me.

Shortly after that, one of the Prairie State Audio Construction Society meetings was held at the home of a member who had a pair of RD75s in a monopole configuration. Although the sound was different, it was a case of same church, different pew. The basic character of the driver was still wonderful.

Attending that meeting was Audio-X-Stream's Rudi Blondia, the distributor of BG products to the DIY market, and also a very knowledgeable and helpful guy. To say that he believes in these drivers is an understatement. Rudi mentioned that he had a pair of RD75s with a custom-designed baffle at his house not far from the Los Angeles airport, and that I should visit him if I were in the area.

MOUNTING METHOD

The question now was not whether RD75s were going to be in my future, but only how. Being a dipole lover, the first part of the equation was easy. The configuration would be dipole. But how would I mount them? Would the baffle be flat, curved, symmetric, asymmetric, or what?

Back to Rudi, or rather his website at www.audio-x-stream.com. He provided not only a lot of information on the drivers, but also links to work done by John Whittaker and himself in 1997, testing various baffles with the RD75 and other planars. Their results confirmed that the best response with the RD75s was obtained by using curved, asymmetrical baffles, a configuration which reduces the tendency for these drivers to have a bump around 300Hz. For more information on that research, consult Rudi's site.

Visions of curved asymmetric baffles began floating around in my head. I needed to narrow the choices. As luck would have it, I was going to LA on business, so I called Rudy to arrange a visit. On my way back to the airport, I stopped by and heard his implementation of the baffle. It sounded great. Then he showed me a new design that was still in prototype stage, using multiple curved sections of plywood.


The completed planar magnetic driver.

Based on Rudi's prototype, I decided on a baffle with a very short distance from one side of the driver to a small-radius curve on one side. The other side of the driver would face a larger baffle made essentially from a 12″-radius curve mating with a 4″-radius curve and then returning to the back side on a straight line.

One of the most significant problems I faced was building a 7′-tall baffle in my basement. The low ceiling height, coupled with HVAC ducts, plumbing, and electrical conduits, would make it very difficult to move large sections without hitting anything. Then good fortune struck. I happened to mention the dilemma to my neighbor who is a building con-

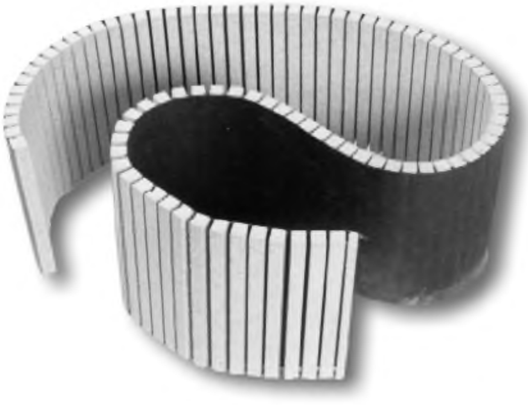


PHOTO 1: Flexible Kerfkore.

tractor, and he mentioned that he had a millworks in town where he produced custom cabinets for some of his jobs. He suggested I contact his manager to see if he could help.

Talk about manna from heaven! I went down to his shop and found that he not only had a large, well-equipped workspace, but that his manager, Dave Coombs, was willing to have me work there. Now I needed to become serious about putting my ideas on paper.

AIRPLANE-WING SHAPE

While making a scale drawing of the cross section of the baffle shape, it struck me that it resembled drawings I had done when building model-airplane wings. Could a speaker baffle be built like a model-airplane wing? Building a wing requires a series of ribs to determine its shape, one or more spars for structural strength, and an outer skin to direct the airflow. The advantage of this method is that you can achieve complex shapes by making a pattern for the ribs and then cutting them out of flat stock.

The process of creating the ribs and spars for the baffle was a simple transfer from the procedure of making a model-airplane wing, although on a larger scale. The difficult part would be finding a suitable material for the rib covering. For an airplane wing, a thin but tough plastic material is stretched and sealed over the ribs and spars to keep the airflow (essentially a steady-state pressure) from passing through the wing.

That kind of material would not do for a speaker baffle, however, where the airflow is not steady-state, but a varying pressure. The electrical analogy is the difference between a DC and AC signal, where the wing coating behaves like a capacitor, blocking DC but passing AC.

Using a wood veneer would not provide a smooth appearance or sufficient structural strength if applied directly to the ribs, and curved plywood sections would not do because of the difficulty of achieving a smooth joint between sections having different radii. I needed to find a new material. Dave, after some checking with one of their suppliers, came up with two possible solutions.

The first was a material called Timberflex®, which consists of a 1/8″-thick sheet of birch plywood attached to a series of either 3/8″ or 5/8″ stringers spaced about 3/16″ apart for the full length of the sheet. This results in a thickness of either ½″ or ¾″, depending on which version is used. The material is flexible in one direction, and relatively rigid in the other. At first it looked promising, but the manufacturer's specs give 5″ as the minimum radius, and the sample supplied seemed more comfortable with a 6″ radius. In either case, it was too stiff to work with the 4″ radius I planned to use.

THE KERFKORE DECISION

The second material, called Kerfkore®, consists of ½″ stringers separated by about 3/16″ like the Timberflex, but attached to heavy black paper instead of plywood. This results in a much more flexible material that bends to a radius approaching 2″. It looked as though I had a winner (*Photo 1*).

I then produced a full-size drawing of the cross section of the baffle, detailing each piece. In constructing the baffle, I would use ¾″ MDF for the ribs, spar, and front driver-mounting plate, and ½″ particleboard for the back plane. I would cover the ribs with Kerfkore, and form the baffle short-end radius with sections of 1 1/16″ quarter-round, since no suitable half-round was available at the local lumber supply.

I would fabricate the top and bottom covers, which would add to the appearance of the baffle and provide mounting stability, from solid planks of hardwood, and the MDF and Kerfkore would be covered with a hardwood veneer. "Roadie box" carpet would cover the backplate and back of the spar. I made multiple copies of the cross-section drawings to facilitate producing the templates necessary for the ribs and the top and bottom plates.

I formed these templates by cutting out the appropriate sections of the drawings, gluing them to a piece of 3/16″ Luan board, and then cutting the board to match the drawing. In addition to providing an outline of the parts to be cut, these templates provided a way to ensure alignment of the mounting holes between the frame and top and bottom plates. *Photo 2* shows one of the templates.

To prove the concept, I next produced a prototype consisting of most of the major pieces. It was made of two ¾″ MDF ribs, a ¾″ MDF spar, a ½″ backplate (particleboard in this case, but it could as well have been MDF), a driver mounting plate made of ¾″ MDF, and a piece of Kerfkore to cover the ribs. The prototype did not have top or bottom sections or the short end radius, as these were relatively straightforward pieces.

The pieces of MDF and particleboard were cut and assembled to form the frame. *Photo 3* gives a view of the general frame construction. I then glued and nailed the Kerfkore to the frame. The concept worked. *Photo 4* shows the completed prototype.

CONSTRUCTION STARTS

I then started to construct the actual pieces. One 4′ × 8′ sheet of ¾″ MDF was sufficient for all the MDF parts. The first step was to cut out ten rib sections. I ripped square pieces just large enough for the ribs from the MDF sheet, and drew the outline for each rib on those pieces, using the rib template. I used a band saw to make a rough cut of the rib shape, and finished the shaping with a large disc sander. The ribs had a smooth curve, terminating at one end in the flat section I

PHOTO 2: Basic template.

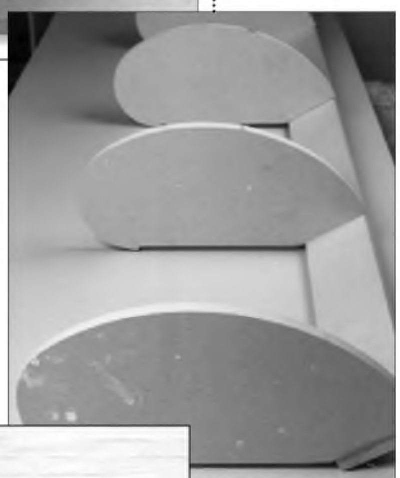
PHOTO 3: Prototype frame.

PHOTO 4: Prototype with Kerfkore.

glued to the spar, and at the other end in a recessed flat section to accept the flat back plate.

Using glue and a pneumatic nail gun, I then fastened the ribs to two spars ripped from the MDF sheet. For those of you who have not used a nail gun, you're in for a real treat. Not only is the assembly a lot faster, but it's a one-handed operation, leaving the other hand free to help maintain alignment during the process. Once the glue is dry, the resulting joint is very solid. To provide additional stability, I used corner blocks where the end ribs joined the spars (*Photo 5*).

▲ **PHOTO 5: Corner braces.**



▶ **PHOTO 6: Ribs connected to spar.**

▼ **PHOTO 7: Dave meets ribs.**

Once I fastened all the ribs to the spar, I could visualize the structure of the baffle a little easier. *Photo 6* is a view of a finished rib/spar assembly. To give you a sense of scale, *Photo 7* shows Dave holding one of the rib assemblies.

The next step was to cut and attach the back plates. I cut these pieces from ½″ stock and rested them in the recesses of the ribs. This combination of plate thickness, recessing, and the thickness of the Kerfkore resulted in the plate being about ¼″ below the edge of the Kerfkore, a difference in height that allowed for the thickness of the carpet to be added later.

The plate joined the spar at an angle of 30°, so I needed to cut one edge of the plate at that angle. I then glued and nailed the plate into place (*Photo 8*). Once the back plate was secured, the whole rib assembly became quite rigid.

MOUNTING PLATES

The only other parts made from MDF were the two driver mounting plates. At first I thought of cutting the plate as one piece and routing the opening, but because of the shape of the driver, that job would have been rather complicated. The openings for the drivers are long and narrow, which would require setting up long guides to make sure the holes were straight. I would need to rout another groove along one edge to accept the Kerfkore sheet. In addition, some raised areas at the top and bottom of the drivers allow for the electrical connections, and the plate would need routing to provide clearance for those areas.

Dave listened to my plan, looked at the drawing and the driver, and said, "That looks too difficult. I think I know an easier way. Let's build it up from individual straight pieces of MDF." He promptly whipped out a pencil and drew four straight pieces of MDF. One was the side of the mount that had the recess for the Kerfkore. This was no problem to make: simply cut a straight piece on the table saw and then make two more passes on one edge with the saw blade and rip fence adjusted to the correct dimensions to produce the relief.

The second piece was the side of the mount away from the Kerfkore, and that was a simple straight cut. The last two

PHOTO 8: Backplate attached.

pieces were the top and bottom sections, which were straight cuts with two additional passes to make the recesses for the raised connection areas of the drivers. Again, it was a matter of two settings of the rip fence and blade height.

Then, to ensure strength in the built-up parts, Dave cut some slots in the edges of the mating parts and made some splines to reinforce the joints. We could have used biscuits, but Dave's biscuit machine was out on a job. *Photo 9* shows the parts before assembly. Note the pencil marks on the pieces, which ensured proper alignment in case the slots for the splines were not exactly centered when they were cut. We then glued and clamped the pieces and let them dry. The results were outstanding, showing what you can accomplish with a table saw if you know what you're doing.

Once the mounting plates were dry, we glued and nailed them to the rib assemblies. The recessed edge of one side resulted in a flush mounting surface with the

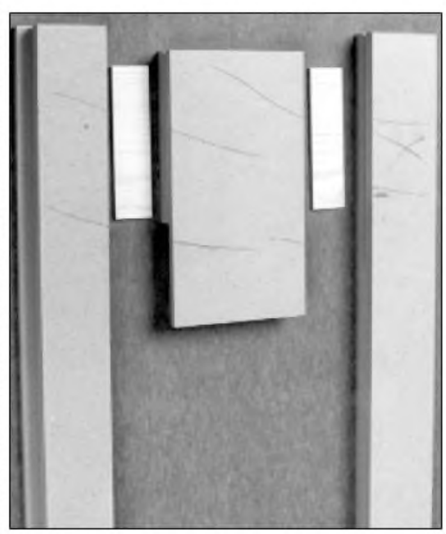


PHOTO 9: Pieces for front plate.

PHOTO 10: Side view of front plate.

PHOTO 11: Front view of front plate.

edge of the rib, allowing the Kerfkore to start at the mounting plate and smoothly transition to the ribs. The relationship of the ribs, spars, backplate, and mounting plate is shown in *Photo 10. Photo 11* is another view of the assembly, with a clearer view of the relief groove in the mounting plate and the relationship to the rib.

TOP AND BOTTOM PLATES

Before covering the ribs, we needed to provide for mounting the top and bottom plates. To fasten the plates to the top and bottom ribs, I inserted ¼″ × 20 T nuts into the ribs to accept corresponding bolts and screws. For connections to the mounting plate and end half round, I would use wood screws, with pilot holes drilled for them. *Photo 12* shows the T nuts in place.

If you have ever used T nuts in a blind hole, you know that there is a risk of pushing the T nut out of the hole while assembling the part. To prevent this, I glued some scrap pieces of wood over the T nuts, which not only anchored them in place, but also acted as a barrier to prevent the foam insulation that would later fill the cavities from filling the threads. *Photo 13* shows those retainers in place.

Next came the moment of truth. Would I be able to get the full sheet of Kerfkore to work the way it did on the prototype? I measured and cut the sheet with about an

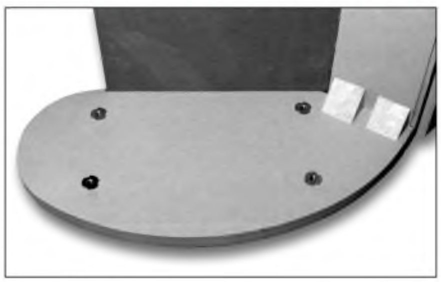


PHOTO 12: "T" nuts in base.

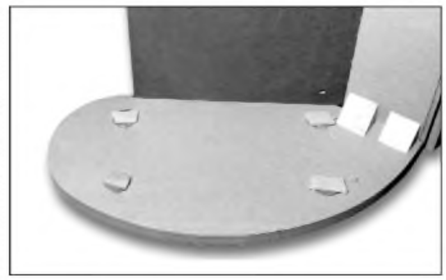


PHOTO 13: Securing "T" nuts.

THE BOHLENDER GRAEBENER COMPANY AND PRODUCTS

Bohlender Graebener (BG) is a company located in Carson City, Nev., that produces high-quality planar magnetic audio transducers. Over 30 years ago, David Graebener and four other speaker enthusiasts started a company called Speaker Lab in the Seattle area. At that time, Mr. Graebener began developing the transducer that is currently in production.

Approximately ten years ago, he set out on his own, completed the development of this technology, and marketed these speakers under the company name of Applied Technology Engineering. According to Ed Dell of Audio Amateur Corp., David's involvement in an earlier company, Speaker Lab, was also the inspiration for Ed to start *Speaker Builder*. We therefore must thank David for more than just producing his line of planar magnetic transducers.

Approximately six years ago, David, Tom Bohlender, and Warren Kocmond concluded that this type of speaker technology had some serious potential in today's marketplace. They gathered a group of sophisticated private investors and formed the Bohlender Graebener corporation for the purpose of further developing and marketing this unique technology. During these six years, the technology has been developed to the level where it can be manufactured in a repeatable and consistent manner, in various forms and sizes, using a variety of materials suited for specific performance requirements.

Today, the company manufactures drivers under OEM contracts for several prestigious loudspeaker firms. In addition, they produce two product lines using their planar magnetic drivers, a line of floor-standing speakers and another of in-wall speakers. Visiting their website, www.bgcorp.com, will give you information on some of the new materials and design concepts they are investigating to further the performance of this technology.

The website also has information on the performance characteristics of this type of driver in general, as well as specific results from the different models. For those interested in the in-wall arrangement, you can download a paper on the procedures to follow.

For the DIY person, BG offers four basic models, the RD75, RD50, RD40, and RD28.1. These model designations indicate the approximate length of the driver. *Figure A* gives the detailed dimensions of the RD75 I used in my project, and it is pictured in *Photo A*. The basic construction of the standard models consists of a frame made from heavy-gauge painted steel enclosing a push-pull array of Ceramic 8 magnetic-alloy field pieces.

The diaphragm resides within the field created by these magnets. It is a thin polyester film with an aluminum voice coil deposited on its surface. This design results in a load to the amplifier that is essentially resistive, with a 6Ω value in the case of the RD75, making it a very easy load to drive for almost any amplifier.

The rated sensitivity of the RD75 is 88dB/1W/1m. This may seem low, but since the RD75 is a very long line source, you are listening in the near field at any distance in a normal room, even one that is quite large. The decrease in SPL with distance is half that of a standard small-diameter dynamic driver.

Therefore, at normal listening distances, the SPL from the RD75 will be higher than an equivalently rated dynamic driver. That's the good news. The bad news is that the decrease in SPL will not match that of a dynamic driver used in the mid- to low-bass region, requiring some adjustment to get proper balance at a prime listening position.

Another advantage of a long, narrow driver is that it produces a cylindrical wave launch. The sound dispersion is very wide in the horizontal direction because of the narrow width of the driver, but is very restricted in the vertical direction because of its long configuration. Unlike small drivers that tend to produce a spherical wave launch, this cylindrical launch minimizes interfering reflections from the ceiling and floor, one of the main causes of room problems. Using the RD series in a dipole format also produces nulls at the sides, eliminating some of the other nasties you can get from early reflections.

The other models are essentially just shorter versions of the RD75. The resulting performance characteristics center on power handling and impedance. For example, the RD75 can handle 200W of program material, while the RD28.1 can handle 65W of program material. This is understandable when you look at the radiating area of both. The RD75 has 144 in^2 of radiating area to the RD28.1's 48 in^2.

Looking at the RD75, you realize that 144 in^2 is not too shabby for a mid/high frequency driver. Your typical 7″ driver comes in around 28 in^2, and although it may have a higher X_{MAX} if it is a combination bass/midrange, I know of no mid/high driver that can match the RD75 in terms of volume displacement. Because of this large area, the RD75 diaphragm moves a relatively short distance even at high volume levels, resulting in lower IM distortion. And the most endearing feature of these drivers is that they cover the mid/high frequency range in one driver, including the all-important human voice range.

The impedance of the RD75 is 6Ω, while that of the RD28.1 is 4Ω. However, since they share most of the same acoustic properties, matching the different versions for applications such as surround sound should be easy. For all the drivers, BG specifies a crossover frequency of 150H at 24dB/octave. For full power handling, they recommend a crossover point of 300Hz. For the technically inclined, the following are some of the materials you can use to fabricate custom drivers:

DIAPHRAGMS:
Polyester
Polyethylene napthaliate (PEN)
Polyamide (Kapton)

CONDUCTORS:
Deposited aluminum
Etched aluminum
Etched copper
Various surface applied wound metal wires

FRAMES:
Carbon steel
Stainless steel
Various molded plastics

MAGNETIC MATERIALS:
Strontium-ferrite ceramic
Isotropic and non-isotropic alnico
Samarium-cobalt
Neodymium-ferrite-boron

That shopping list should keep you busy for a while. However, for general home applications, the standard versions as supplied to the DIY market work just great!

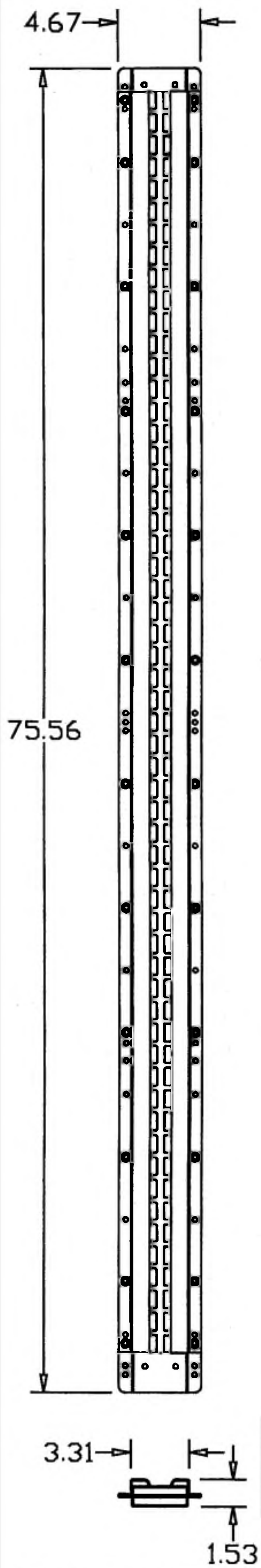


FIGURE A: Detail dimensions for the RD75 driver.

PHOTO A: The Bohlender Graebener RD75 driver.

inch of excess beyond the top and bottom ribs, placed it on the rib assembly, and checked it for fit. Then I applied glue to the mounting-plate recess and rib edges, and nailed the Kerfkore to the mounting plate and the leading edges of the ribs. Finally, I wrapped the balance of the Kerfkore and nailed all the way around. *Photo 14* shows the operation in progress.

After allowing the glue to dry, I trimmed the excess Kerfkore using a router, which resulted in a nice flush edge for mounting the top and bottom plates. I built up a half-round edge from two quarter-round pieces and glued and nailed it to the other edge of the mounting plate. (Normally, I would have preferred to purchase the half round, but no suitable sizes were available locally.) I then drilled holes through the back plate on both sides of the rib cavities to allow for inserting sound-absorbing insulating foam at a later stage. *Photo 15* shows the completed rib assembly.

WOOD SELECTION
I now needed to make a major decision before I could fashion the top and bottom plates. Up to this point, all the wood was generic, but since I wished to make the top and bottom plates of solid hardwood, the choice of wood and veneer type was at hand. I spent a lot of time looking at various rare woods, but the final decision required that the wood be available and fit with current other pieces of furniture in my home.

Having for a long time been a fan of walnut, I picked American walnut for its color, grain, and ability to fit with current pieces. However, finding local planks large enough for these plates proved to be impossible. Instead, I settled for some very nice pieces of 1″ planks in various widths and lengths out of which to build up larger pieces.

The following are points to consider when building pieces such as this. Use raw stock pieces that are complementary in color and grain, making sure the edges are very square, straight, and smooth. Reverse the grain patterns when viewed end on from grain up to grain down on adjacent pieces. Since good hardwood is quite expensive, you should avoid making square pieces if the shapes you desire are not square. Rather, leave only enough excess to make the assembly and cutting operations easy, with a minimum of waste.

I examined pieces of raw stock, cut them to length, and placed them next to each other to make sure all was OK. Then I made pencil marks across mating sec-

PHOTO 14: Applying Kerfkore.

PHOTO 16: Built up base stock.

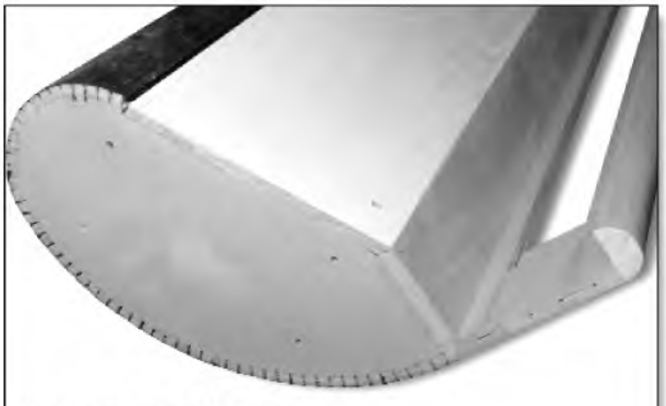


PHOTO 15: Completed wing.

PHOTO 17: Base stock with template.

tions to assist in reassembling them properly, and ran them through a jointer until smooth and straight. I applied glue to the mating edges and clamps to hold them in place. I clamped them lightly at first, and used a hammer and wooden block to ensure all edges were flush with each other. When alignment was complete, I further tightened the clamps and allowed the glue to dry.

Do not overglue when working with bare hardwood, because excess glue that seeps onto the surface of the wood will interfere with the finishing process. Apply clamp pressure slowly, looking for any glue seepage. Immediately clean any that appears before continuing the clamping. *Photo 16* shows one of the assembled hardwood pieces.

FITTING BASE AND TOP

To cut the base and top pieces, I used templates, making the original drawings on Luan board as for the ribs. For stability at the bottom, I added an extra 4″ beyond the frame dimensions, and did not draw the rear of the base to follow the baffle, but rather went more directly across the back, leaving room to mount a terminal block. *Photo 17* shows the bottom template on a hardwood piece ready for marking.

After marking the outline of the piece, I drilled pilot holes and cut the piece using a band saw. I did the final shaping with a disk sander, and routed the edges for a rounded look. *Photo 18* shows the bottom piece after cutting, pilot-hole drilling, and edge routing.

For the top, I cut the template to fit the frame. I cut a small wooden block as a guide, with a hole drilled 1″ from the edge. With a pencil inserted through the hole, I used this to trace an outline 1″ beyond the edge of the baffle, except for the rear, where I left additional material, as with the bottom. I drilled pilot holes for mounting and cut and routed the piece. *Photo 19* shows the result.

To provide a secure and attractive mount for the electrical connections to the speakers, I decided to make two terminal-mounting boxes out of the scrap pieces left over from the bases and tops. These boxes would hold the heavy-duty binding posts I chose. As in the past, I decided to use Vampire connectors, since they have provided the best quality connectors at the most reasonable prices.

For this project I used two pairs of the BPHEX connectors. It is very annoying to have your binding posts rotate when you are tightening them, but the Vampire posts are inserted into a base that prevents this. Part 2 will describe construction of the enclosure and the proof of the pudding—listening tests. ❖

PHOTO 18: Cut and drilled base.

PHOTO 19: Cut and drilled top.

The DR10 Horn

The author's latest series of horn-loaded cabinets is dubbed the Double Reverse (DR) line and is guaranteed to generate lots of interest.

By Bill Fitzmaurice

I read Louis McClure's "Exponential Mid-Range Horn" articles (*SB* 7/99 and 8/99) with great interest. It's nice to see someone else challenging prevailing notions about horns.

For those of you who missed these articles, McClure built two midrange horns of similar length and mouth area, but with differing throat sizes and taper rates. The nearly identical response curves of the two horns defy traditional theory. Listeners' very different perceptions of the sound quality of the two horns, despite similar response curves, were more puzzling to McClure. He finished his experiments with perhaps more questions than answers.

I found his results confirm my own theories that the passband of a horn is mostly a factor of its length, efficiency is decided by mouth area, and that a tighter throat will give better high-frequency loading. While the passband and efficiency of McClure's two horns are very similar, the smaller throat version has a decided 6dB hump between 500 and 800Hz. If you'll recall the Fletcher-Munson curves, this is where the human ear is most sensitive. With a peak in that area I would expect it to sound harsh, as McClure described, especially when not sonically counterbalanced by any fundamental tones below 200Hz. How curious it is that those frequencies we are most sensitive to are also those that grate the most on our nerves, causing us to desire more and more bass.

PHOTO 1: Test box w/10″ driver.

The quest for bass brings me to my own experiments in throat design. In minimizing throat sizes for better midrange loading, I found that smaller throats also result in lower f_B. To better understand the relationships between throat size and f_B, I built a 1ft^3 test box housing a cheap 10″ driver (*Photo 1*). To this box I attached a horn.

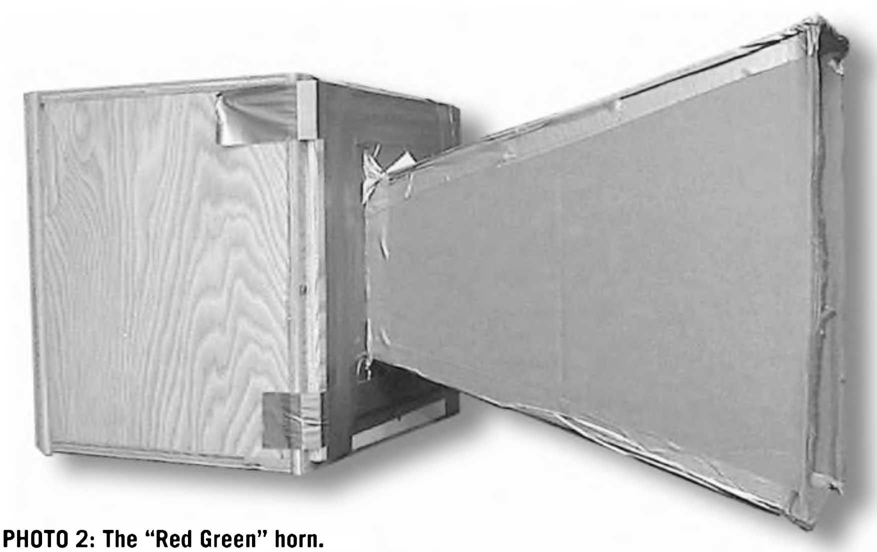


PHOTO 2: The "Red Green" horn.

Unlike McClure's magnificently constructed horns, however, my construction techniques came straight from the "Red Green Show": cardboard and duct tape (*Photo 2*). I charted the resonant frequency of the unaltered box and a variety of horn configurations. The results are displayed in *Table 1*.

ABOUT THE AUTHOR

Bill Fitzmaurice (BA., University of New Hampshire) has been a professional musician since 1966 and has been constructing instruments, amplifiers, and speakers for just as long. As a resident of Laconia, NH, since 1981, he spends his winters skiing, his summers playing golf, and his weekends playing electric bass and running sound for the "L.A. East" rhythm and blues band, of which he has been a member for 15 years. Owner of L.A. East Recording Studios, he has also managed to find time to write an adventure novel. Fitzmaurice is the author of "Loudspeakers for Musicians" and over 30 magazine articles dealing with speakers and electric instruments.

TABLE 1
SYSTEM SPECS
RESONANT FREQUENCY OF THE UNALTERED BOX AND A VARIETY OF HORNS

CONFIGURATION	THROAT AREA	LENGTH	MOUTH AREA	RESONANT FREQUENCY
Box Only	n/a	n/a	n/a	110Hz
Horn	36 in^2	16″	96 in^2	95Hz
Horn	16 in^2	16″	96 in^2	65Hz
Horn	16 in^2	8″	40 in^2	70Hz
Horn	16 in^2	4″	28 in^2	83Hz
Constricted Baffle	16 in^2	n/a	n/a	100Hz

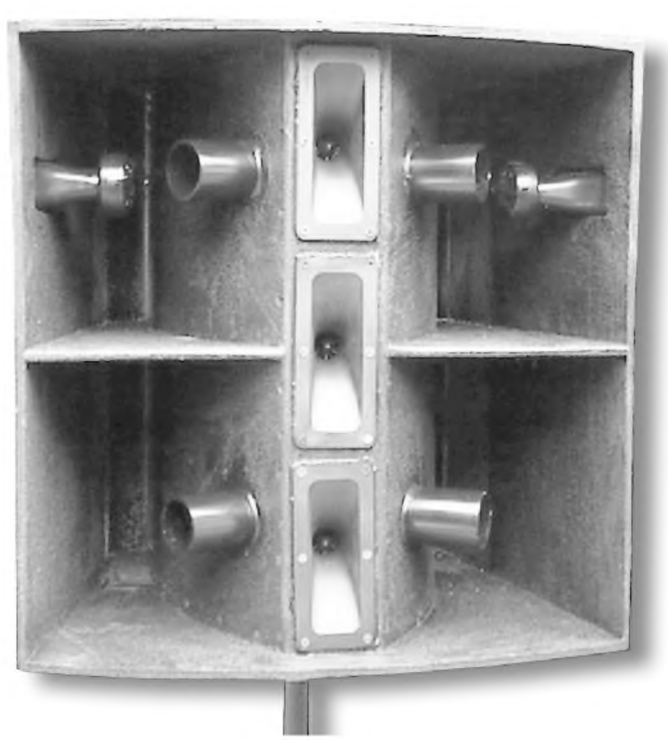


PHOTO 3: The DR10, front view.

PHOTO 4: The DR10, side view. Note optional side-fire tweeter.

The results were a revelation to me. Obviously, the size of the throat is more important than either the length of the horn or the area of the mouth in lowering the system resonance. I was most surprised by the constricted baffle configuration, which was simply a plate with a hole in it, placed over the driver opening in the baffle. Why did this cause a 10Hz drop in the box resonance? I realized that the space between the driver cone and the constricted baffle opening became a resonant chamber, turning the closed box into a simple dual-chamber reflex box.

How does this relate to horn design? Consider a horn/reflex cabinet with a desired low frequency cut-off at 40Hz. Maximum efficiency is achieved if the driver/horn resonance (the f_S of the driver after mating it to the horn, but prior to enclosing the rear of the driver) is close to 40Hz. If the driver/horn resonance is significantly less than 40Hz through use of too small a throat, then mid-bass performance will suffer.

On the other hand, too large a throat will result in poor midrange loading, as well as too high a system resonance. Clearly, the proper size throat opening is critical for maximum performance. With that fact in mind, I spent more time working on the throat configuration of my latest design than on any of my previous cabinets. With the results I achieved I think I am on the right track.

[Note: I have not seen described a term which denotes the resonant frequency of a driver mated to a horn while the rear of the driver is exposed to free-air, so I have coined one: f_S(h)].

HORN DESIGN

The impetus for this cabinet was an inquiry from a manufacturer interested in my Snail folded horns for bass, keyboard, and PA. [For a collection of the author's work with Snail folded horns, see *SB* back issues and *Loudspeakers for Musicians*, available from Old Colony Sound Lab, 888-924-9465 or custserv@audioXpress.com.—Eds.] He had three design requirements. First, the design needed to be compatible with molded construction, most likely fiberglass. Second, the design must be suitable for medium-priced drivers, rather than the premium EVs™ and JBLs™ I'd used in most of the Snail designs. Last, the design needed to work full-range as a two-way system, preferably using piezo drivers for the high end. (It also wouldn't hurt if it blew away anything else currently available on the market.)

The result of my efforts to meet these requirements is the 10″ loaded DR10, the first in a series of cabinets using drivers from 8 to 15″ (*Photo 3*). It can be duplicated in fiberglass, or another molded construction method, by producing sub-assemblies (the throat horn, the mouth horn, and the exterior shell), which would be bolted together. To make cheaper drivers feasible, the cabinet is larger than a Snail designed for a ten would have been. This allows a longer horn path and larger mouth for better efficiency over a wider bandwidth; the mid-line driver-equipped DR10 outperforms the smaller Snail and MidRanger, despite their being loaded with premium twelves.

The mouth opening is square, which is more efficient than a rectangular one. The throat section is subdivided to minimize its cross-section for better mids, while the mouth is subdivided to provide bracing between the horn and the exterior shell. Unlike traditional designs, the horn taper is not a constant formula from throat to mouth.

Waves from the driver first pass through a rapidly flaring midrange horn, then reverse direction to pass through the mouth horn, which has a slower flare rate. In other words, this cabinet consists of two separate and distinct horns, which are coupled in series: a double horn. Where the two horns join together, the direction of the wave path reverses, thus, the Double Reverse, or DR, horn.

Unlike the Snails, this cabinet does not have driver chamber ducts exiting into the horn throat. While that technique works well with shorter horns, doing so with a longer horn drives f_B too low. Instead, four ducts exit the driver chamber very close to the horn mouth, allowing for easy and precise tuning of the system. As in the Snails, all interior reflectors are round to minimize phase cancellations.

The cabinet's exterior shell also has a trapezoidal shape. I included this feature strictly for cosmetic value, since I once

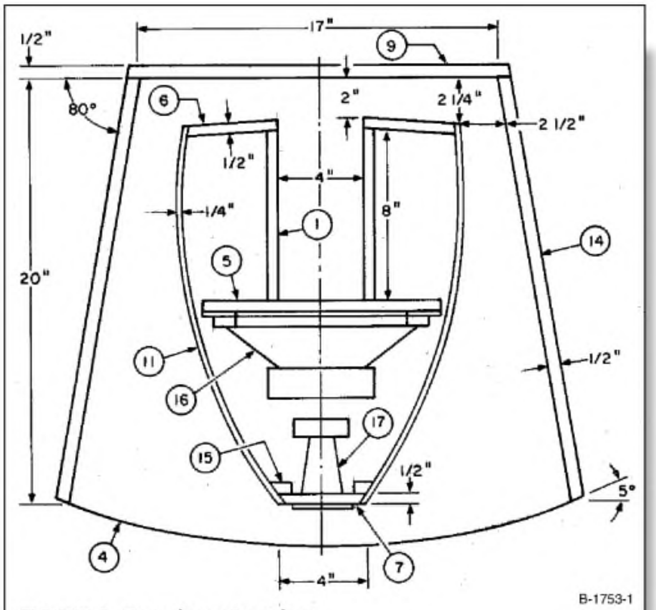


FIGURE 1: Top view of cabinet.

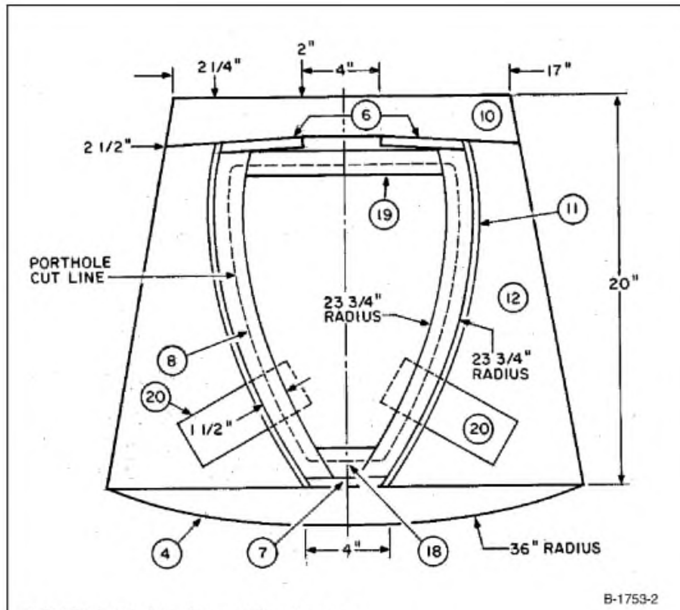


FIGURE 2: Guide for cutting braces.

overheard a fellow musician incorrectly attributing the quality of a commercial cabinet to its trapezoidal shape. A trapezoidal shape in and of itself indicates nothing of a speaker's worth, but if it sells, then why not do it? The cabinet also features an arced leading edge at the top and bottom. This will, in theory, make the box work better, but I included this feature primarily because it looks cool.

The tweeter section consists of three Motorola™ KSN 1176 piezos in a vertical array, for maximum horizontal dispersion. These cut out below 2.5kHz, but the response dip between the woofer and tweeters is not severe, since the woofer is effective to 2kHz. Different drivers could eliminate the dip entirely. The woofer I used—because I had it on hand—is an Eminence™ 10″ cast-frame MI driver that is no longer being produced. Its f_S of 55Hz, Q_{TS} of .25, V_{AS} of 1.85ft^2, and SPL of 98dB are fairly typical of medium-quality MI tens.

In choosing a driver for this project, stay away from premium drivers such as JBL™ or EV™ (which have very large magnets), since they might not fit into the driver chamber. A magnet weight between 40 and 80 oz will do nicely. The most important spec is the f_S; do not use a driver with f_S less than 50Hz or more than 60Hz. The prototype also features optional KSN 1176s on each cabinet side for side-stage projection. Note that these are aligned horizontally, since they provide better horizontal dispersion this way when used singly.

CONSTRUCTION

I constructed the cabinet for the prototype of ½″ plywood, excepting horn

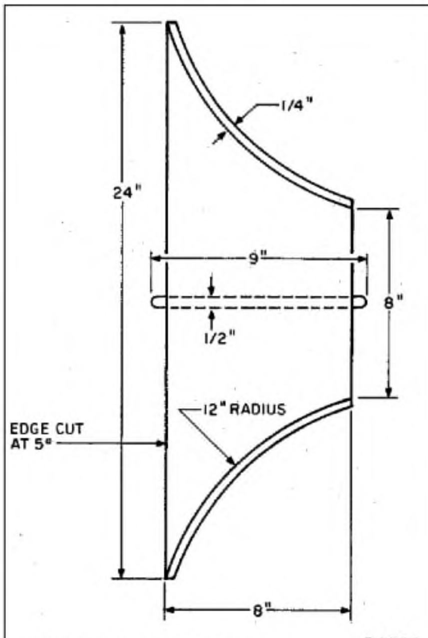


FIGURE 3: Side view of horn.


PHOTO 5: Cutting angled piece of plywood to modify panel cutting jig.


PHOTO 6: Cutting cabinet top using modified panel cutting jig.

sheathing (¼″ plywood). You may use thicker materials for ease of joinery, but rest assured that additional weight is not necessary for strength. The self-bracing design may be lightweight, but it is rock-solid.

Glue and seal all joints with construction adhesive, and secure them with 1¼″ drywall screws driven through drilled and countersunk pilot holes. For thicker materials (I wouldn't recommend more than [illegible]/8″ plywood), use 1[illegible]/8″ screws. Note that all measurements are nominal, the exact dimensions being dependent on the actual thickness of the materials used. Don't try to cut parts much in advance of assembling them; actual part sizes should be double-checked using the previously-assembled parts as a reference.

Before starting construction, thoroughly examine all of the figures and photos. Note that *Figs. 1*, *2*, and *4* are variations on the same view; to include all the components in one view would be too cluttered. *Figure 3* shows a cutaway of the throat horn. The optional side-fire tweeters (*Photo 4*) are not included in any of the figures, nor is any hardware shown. As in all successful projects, you should understand what you're trying to build before you make any sawdust fly.

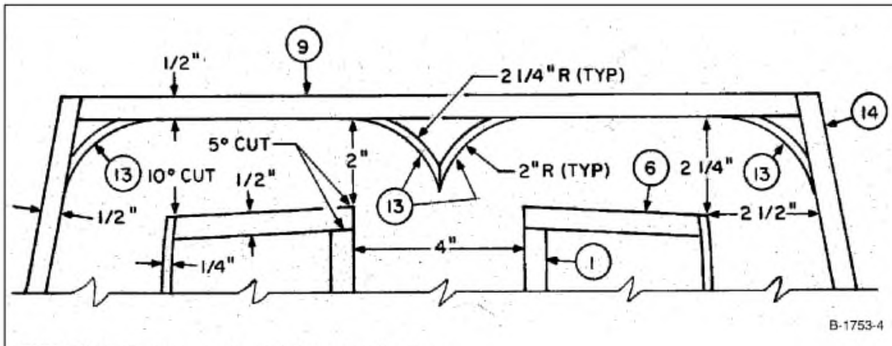


FIGURE 4: Cutaway view of throat reflectors.

MAKING THE CUTS

The first step is to cut out the top and bottom (*Fig. 1*), which requires straight, accu-


PHOTO 7: Checking brace patterns against cabinet top.

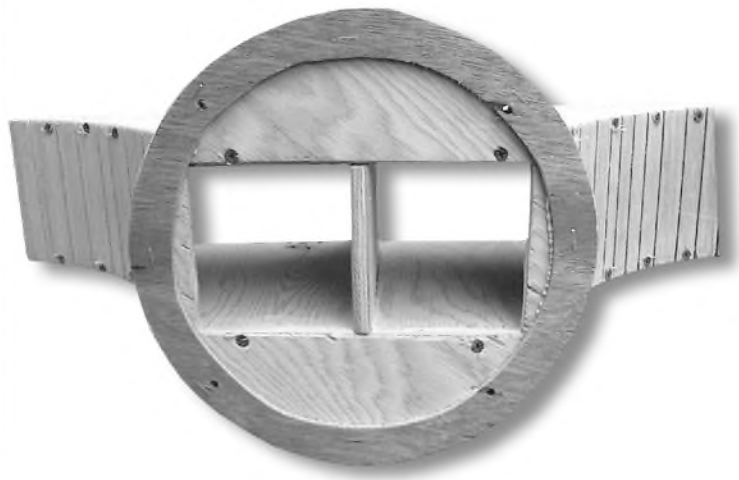


PHOTO 10: The completed throat horn. Note the ¼″ plywood spacer on the face of the baffle.


PHOTO 8: The throat horn sides and throat divider assembled.

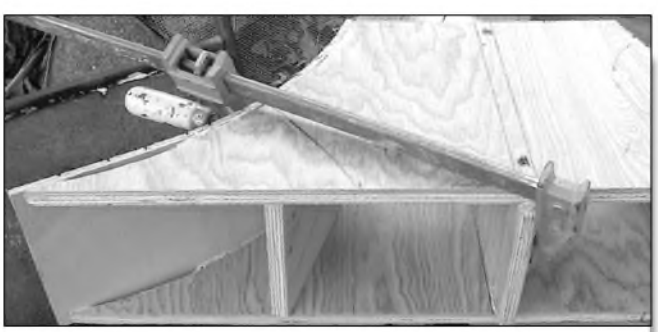


PHOTO 9: Sheathing the throat horn.

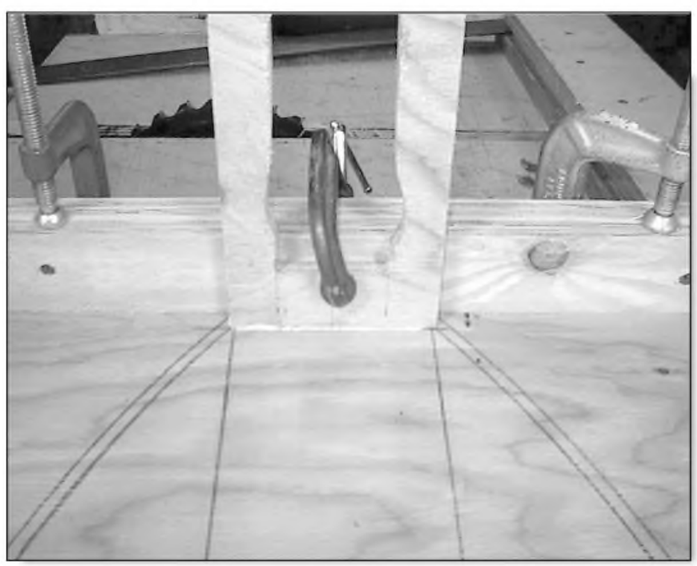


PHOTO 11: Attaching the tweeter baffle to the top using a guide-board for accuracy.

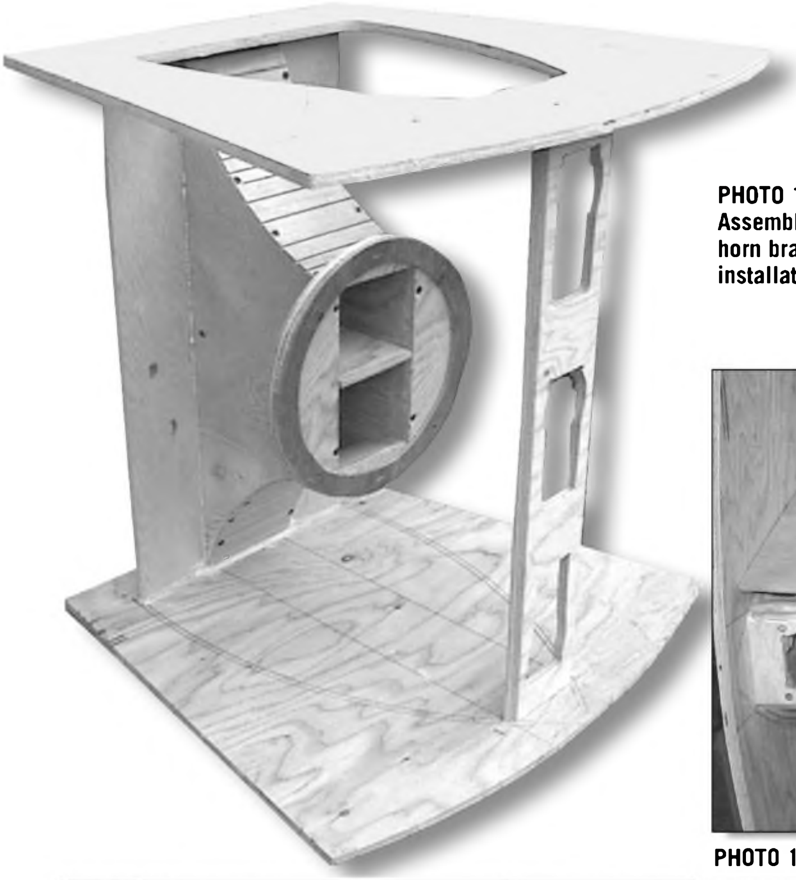


PHOTO 12: Assembly prior to horn brace installation.

rate 80° cuts (or 100°, depending on point of view), instead of the usual 90°. Doing this on a table saw is easy if you first cut two pieces of scrap plywood at a 10° angle, using your miter gauge (*Photo 5*). Screw these directly to your panel-cutting jig (if you haven't yet made one, do it now), allowing you to feed a panel across the saw table at a perfect 80° angle (*Photo 6*).

To ensure that the top and bottom are identical, first rough-cut two pieces of ply-

PHOTO 15: Attaching the mouth horn sheathing.

PHOTO 13: View through the porthole.

PHOTO 16: Installing the back brace.

PHOTO 14: Horn braces cut in two pieces with stabilizing cauls attached.

PHOTO 17: Slicing PVC pipe on the jig.

wood slightly oversize, and then screw the two together with 1″ screws. Now cut both pieces to finished size simultaneously. This trick is valuable anytime you need identical parts.

After cutting the top and bottom, draw the placement of all mating parts onto them (*Figs. 1* and *2*). You can draw the arcs with an oversized compass, a long plywood sliver with a drywall screw at one end serving as a pivot and ¼″ holes drilled through the plywood at the required radius distances from the screw, through which you insert a pencil to draw the arc. Once the parts locations have been laid out, make the patterns for the horn braces, back brace, and side braces (*Fig. 2*), comparing them to the layout drawn on the top and bottom for accuracy (*Photo 7*).

Note that the horn braces are cut to a 23¾″ radius on both the inside and outside, so that, when cutting out a number of braces, the inner edge of one is the outer edge of the next. I made the prototype using ½″ plywood for all braces, but this is one place where thicker materials, even 5/4 lumber, will make joinery easier without much of a weight sacrifice. Mark the cut line on the bottom piece for the driver access porthole, cutting it out using a saber-saw (starting with a plunge cut).

Next, lay out and cut the horn throat sides, which have a 5° angle on one edge (*Fig. 4*). Attach them to the throat divider (*Photo 8*). The divider extends ½″ beyond the side edges; round both its ends over, either by sanding or routing. Using clamps and one or two plywood scraps, cut to 4″ wide, as temporary braces to hold the throat sides square precisely 4″ apart, while attaching both the divider and the throat sheathing (*Photo 9*).

The sheathing, cut from ¼″ plywood, must be kerfed on the table saw with cuts about ⅛″ deep across the panel at about 1″ intervals, which allows it to be easily bent to the arc of the sides. Cut the sheathing a bit longer than required, with the ends trimmed or sanded to exact size after assembly for a perfect joint with the baffle and cabinet assembly.

ACCOMMODATING THE DRIVERS

The baffle is a circle of plywood 11″ in diameter through which is cut a 4″ × 8″ rectangular hole, centered on the baffle's axis. Attach a spacer ring of ¼″ plywood to the baffle to prevent the cone edge from slapping the baffle in long excursions. Drill the baffle and insert four T-nuts into it for driver mounting (using only every other hole on the driver) with

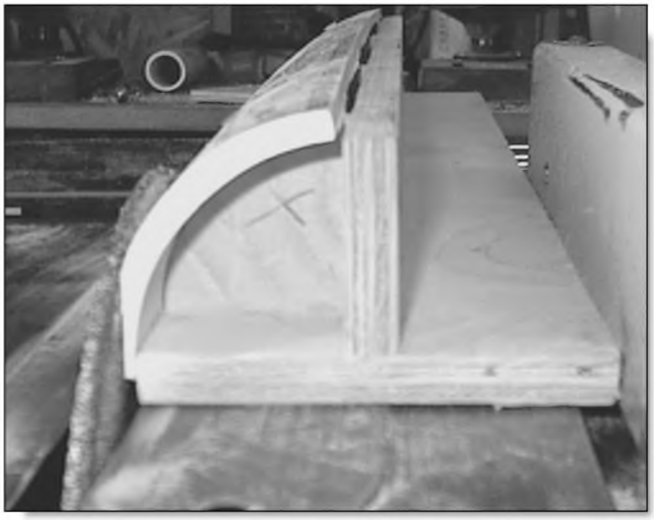


PHOTO 18: Chamfering quartered PVC on the jig.

PHOTO 19: PVC throat reflector installation.

the holes aligned near the corners of the hole, not along the sides. Attach the baffle to the throat assembly, after sanding over the hole opening to eliminate sharp edges (*Photo 10*). Where the baffle meets the throat sheathing no screws can be used, so use plenty of adhesive for an airtight fit.

Cut the tweeter baffle to size, and then cut holes for the tweeters through it, equally spaced across its length. The cut angle for the tweeter baffle sides is 35°. If you use ½″ plywood for the baffle, cut some extra strips of plywood at 35° angles as bracing blocks, which will later be attached to it. For thicker materials, additional bracing will not be necessary.

The holes for KSN 1176 frames measure 5⅛″ × 2⅛″, but the driver element housing will not fit through holes of that size, so trim a bit of material, as required, from the holes to allow the drivers to fit. Attach the baffle to the cabinet top. This (and all similar joints) is best accomplished by first drilling pilot holes through the top along the joint center and then clamping a straight guideboard to the top along the joint line.

Apply adhesive to the edge to be mated, and clamp the baffle to the guideboard (*Photo 11*). Only when you are sure that the joint is perfectly aligned should you drill through the holes, using a pilot/countersinking bit, and screw the parts together. Using one drill for piloting and another for screwing makes this a fast and easy procedure.

Cut the horn plates to size, paying close attention to the angles of the edges (*Fig. 4*), and in similar fashion attach them to the top. When the cabinet bottom is complete, attach it to the assembly. Trim and sand the throat assembly for a proper fit with the rest of the cabinet and secure it in place, again using no screws and plenty of adhesive where the horn sheathing mates the cabinet top and bottom (*Photo 12*).

Tracing from the patterns, cut seven more horn braces, for a total of eight, and one more side brace. Attach the horn braces to the assembly, with four attached to the top and bottom and the other four spanning from the sides to the tweeter baffle (*Photo 13*). On the bottom, use plywood to fill the gaps between the braces (*Fig. 2*), forming a mounting flange for the porthole.

Place the spanning braces about 6½″ from the top and bottom. If placed too close to the baffle, they will interfere with accessing the driver attachment bolts; if too close to the top and bottom they will interfere with the ducts. Cut the two spanning braces closest to the porthole into two pieces to allow the driver frame to pass between them.

Use temporary cauls, screwed to the braces, to hold the braces in position until after the horn sheathing is in place (*Photo 14*). You may need to do some additional trimming to these braces, depending on the size of your driver magnet. Do this now, making absolutely sure that you can fit the driver through the porthole and slide it into position on the baffle. Once the sheathing is in place further trimming is very difficult—which I can attest to from personal experience.

Cut the bracing strips (for a ½″ tweeter baffle only) to length and install them between the horn braces. Cut the horn sheaths to size. The radius of the horn is slight enough that kerfing here is not necessary, but check the plywood for its easier bending axis before cutting the parts. Cut the sheaths a bit long, to be trimmed after installation.

The duct holes, 2½″ in diameter, should be cut with a hole saw, which can tear up thin plywood when unsupported. Drill these before installing the sheathing (with a supporting piece of scrap behind the sheath to eliminate tear-out). The centers for the holes are 4″ from the leading edge of the sheaths and 4½″ from the top and bottom; double-check to make sure the duct holes don't hit the braces.

Attach the sheathing first to the horn plates, then pull it into place with long clamps. At the same time, drive screws every four inches or so (*Photo 15*). Once you have attached both sheaths and the adhesive has dried, trim and smooth the joints with the tweeter baffle, and remove the cauls on the braces.

Attach the back brace to the horn plates and throat divider, using a clamp and two pieces of scrap to align it with the throat divider while driving screws into it from the cabinet interior (*Photo 14*). The rounded edge of the throat divider serves as a trough for adhesive; after it has set, sand the joint smooth. In a similar fashion, clamp the side braces to the back brace while driving screws into them from inside the cabinet.

There is little room inside the cabinet, so these screws must be driven by hand (God forbid!), using a short shaft screwdriver or a small ratcheting driver. This joint is likely to be ragged, and you probably will only be able to get a couple of screws into it, so use plenty of adhesive on the joint to seal the gaps. After the adhesive has set, install the cabinet sides.

WORKING WITH PVC

Cut the throat reflectors (*Fig. 4*) from 4″ Schedule 40 PVC. This stuff is very easy and safe to work with if you use it properly. First, make a cutting jig (*Photos 17* and *18*). Screw the PVC right to the jig using drywall screws. Then push the jig through the saw, using the saw rip guide to set the proper distance from the blade. PVC will bind on a blade if cut all the way through, so set the blade height to cut not quite all the way through; finish the job with a utility knife.

You need to quarter a 24″ piece of PVC; after quartering, screw the pieces to the opposite side of the jig for chamfering. You can use a toothed blade for both cutting and chamfering, but an abrasive blade is safer and cuts cleaner, as well. After chamfering one edge of the PVC, remove it from the jig and reattach it to chamfer the remaining side. Use a T-square to check the angle of the second cut. You'll need two pieces cut at 90° angles, and two at about 100° angles, to fit the 10° flare of the cabinet sides with respect to the back.

Cut the PVC reflectors to length to fit the mounting locations. Use a hot-melt glue gun to attach the 90° pieces to the cabinet, back-to-back, where the throat opens into the cabinet, and the 100° pieces where the sides will meet the back (*Photo 19*). Cut the back piece and install it, using plenty of adhesive where it meets the PVC reflectors; do not drive screws there. If you plan on putting your jack on the cabinet back, cut a hole in the back for it. Cut another hole for the wire to the driver through a horn plate, running a piece of wire through that hole and sealing it with hot-melt glue before attaching the back.

ASSEMBLY

Using a sander and/or router, trim all exterior joints flush and apply your finish of choice. If you plan to attach the porthole with screws, put it in place and drill pilots for them, about a dozen or so. Alternately, you can drill for bolts and install T-nuts on the bottom.

Vacuum the cabinet clean and install the woofer. Because the mounting bolts are difficult to access, it's best to use either Phillips or Allen head bolts, which are easier than slot-head bolts to drive by feel. Again, a short-handled driver or a small ratchet driver works well.

Install the tweeters, wiring them to each other in parallel, and then wiring them in parallel to the woofer. They

TABLE 2
PARTS LIST (ALL SIZES APPROXIMATE—SEE TEXT)

1. Throat Horn Sides	8″ × 24″
2. Throat Horn Divider	4″ × 9″
3. Throat Horn Sheathing	5″ × 12″
4. Top, Bottom	22″ × 24″
5. Baffle	11″ diameter
6. Horn Plates	4″ × 24″
7. Tweeter Baffle	4″ × 24″
8. Horn Braces	4″ × 18″
9. Back	18″ × 24″
10. Back Brace	2½″ × 17″
11. Mouth Horn Sheathing	18¾″ × 24″
12. Side Braces	10″ × 18″
13. Throat Reflectors	24″ × 4″ PVC
14. Sides	20½″ × 25″
15. Tweeter Baffle Braces	(see text)
16. Woofer	10″ MI
17. Tweeters	
18. Porthole Flange A	2″ × 4″
19. Porthole Flange B	1½″ × 10″
20. Ducts	5″ × 2″ PVC

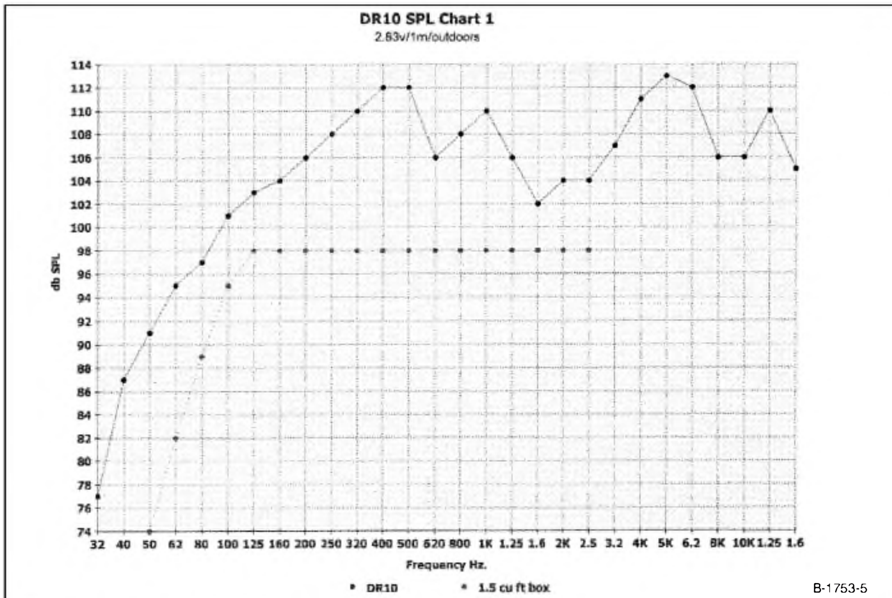


FIGURE 5: DR10 SPL chart.

should be wired in-phase with the woofer. If you decide to use side-firing tweeters, cut holes for them in the cabinet sides and install them, drilling holes through the horn sheath to wire them up, and sealing the holes with hot-melt glue.

The ducts (for a driver f_S of 55Hz) are 5″ lengths of 2″ PVC; install them now, using hot-melt glue to seal them in place. Be careful not to push the ducts in too far because they will hit the tweeters. Spray-paint the ducts to color-match the cabinet finish, and caulk the tweeter frames tight.

Fully stuff the cabinet with at least a pound of poly-fill, being sure to leave no voids except directly behind the woofer and near the duct entryways. For PA usage this box should be pole-mounted for proper projection, making the porthole cover a viable location for both the jack and the pole mount. Alternately, the jack may be side mounted. This will require a hole through the horn sheathing (caulked tight of course) for the wire.

If you're using a pole-mount top hat, the center of gravity of this box is almost precisely in the cabinet middle. Install the top hat and jack, caulking them airtight. Solder the wiring to the drivers.

Rim the porthole flange with ½″-thick neoprene weather-stripping and install the porthole cover. You may attach handles using bolts and T-nuts, but I used screws, driving them through the sides into the side braces. While reasonably light at about 50 lbs, the bulk of this box will make casters a worthwhile option; attach them with bolts and T-nuts. Now plug it in and rock.

PERFORMANCE

If this is your first horn-loaded cabinet you'll be amazed at how loud it is without needing a mega-watt amp to drive it. If you're a bass player you may feel (not just hear, but actually feel) low tones that you've never experienced before. If you're used to playing keyboards through guitar amp speakers, be prepared to hear those tweeters cranking out harmonics you never knew your keys had in them, as well as low tones reminiscent of a pipe organ. Be it on bass, keyboards, or PA, you now have one of the best cabinets available anywhere, at any price, although this cabinet can be built as described for less than $175 per copy.

To back up these statements, look at the SPL charts (*Figs. 5* and *6*). In *Fig. 5*, the dotted trace is the projected response from the same Eminence™ woofer in an optimally tuned T/S box (and keep in mind that this driver is considerably better than most OEM drivers). On average, SPL from 125Hz to 2kHz is about 98dB; with 150W input, output is about 118dB.

On PA, this would work for a small club, but that's about it. For electric bass, the 40Hz to 80Hz (the first octave on the bass) average SPL is a meager 78dB. To get 108dB output, a conservative requirement for bass, would require over 1kW.

Your amp probably doesn't have that much juice available, but that's a moot point, since even four tens couldn't take that much power. Using two tens wired in parallel, or four wired as series/parallel pairs, will produce another 6dB sensitivity, but 84dB sensitivity still doesn't cut it. The only way you can get the bottom you desire using tens in a T/S box is to use eight of them and hope that your amp is big enough.

Now look at the solid trace, the DR10. With average SPL from 125Hz to 2kHz at 107dB, a 150W amp will give 128dB output with headroom to spare. In PA, a pair of these will easily handle a 500-seat room. For the bass player, on average, SPL from 40Hz to 80Hz is 92dB; most T/S boxes with eighteens can do no better. 108dB output requires less than 50W. If you love the tone of your old blackface Fender™ Bassman, but it just isn't loud enough, this cab will cure the problem. On keyboard, the combination of high sensitivity and wide bandwidth gives performance that no commercial keyboard cabinet can approach.

Speaking of commercial cabinets, look at *Fig. 6*. Here the dotted trace on the chart is for a commercial 2-way cabinet using a 15″ woofer and horn loading on

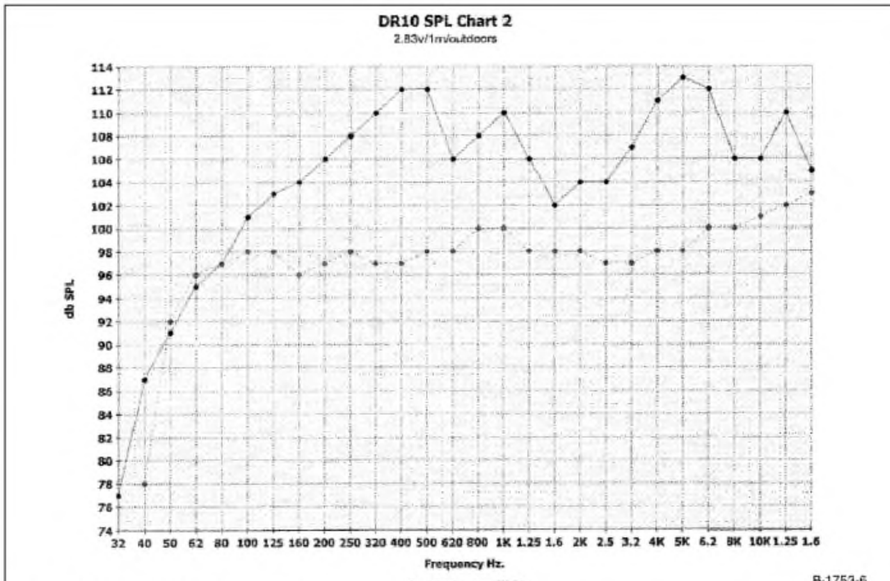


FIGURE 6: EV T251 SPL chart.

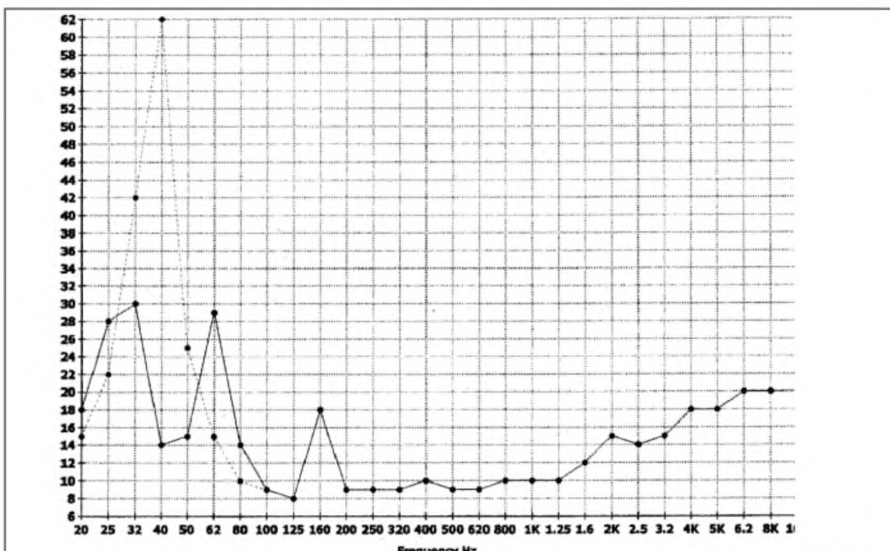


FIGURE 7: Woofer/horn impedance.

the mids and highs, the renowned EV™ T251. It is close to the DR10 in size (about 7ft^3), though at 78 lbs, it weighs over half again as much. Typical of commercial cabinets, it has an average SPL of 98dB.

It will handle 400W RMS input, and you'll need it, for even with over twice the input power it still lags behind the DR10 in output by 6dB. That means you'll need two of them to match one DR10, plus 800W to drive them. Oh, and they retail at over a grand apiece, though you can find them discounted at around $700–what it will cost you to build four DR10s.

DISPELLING MYTHS

Like McClure's midrange horns, the DR10 proves that some assumptions about horns don't ring true. On the subject of throat size, a popular formula states that $St = .8 \times f_S \times Q_{ES} \times V_{AS}$. For the subject Eminence™ driver this would result in a throat of about 22 in^2. Is this correct? I beg to differ.

The dotted trace in *Fig. 7* shows the impedance of the woofer/horn before sealing the box. The target f_S(h) of 40Hz is achieved with a throat of about 30 in^2. A smaller throat would result in an f_S(h) lower than 40Hz, which would give a response dip in the second octave (80-160Hz).

Better high-frequency performance would doubtless accompany a smaller throat, but since we're trying to maximize the bass, the point again is moot. This throat formula may be accurate when used in a compression horn, where the driver is in a sealed chamber small enough to drive the f_B up to the desired frequency, but as far as a horn/reflex combination is concerned it does not work. I do not have a formula to replace it. I arrived at the 40Hz f_S(h) only through a lot of experimentation with a lot of plywood.

A formula to predict throat size for a given f_S(h) would need to include not only the driver parameters but also the horn taper, horn length, mouth size, and the volume of air between the cone and baffle. To calculate it may be impossible. This could be one area where empiricism will continue to rule.

Speaking of formulas, I recall a reply to a reader's letter a few years back questioning how to tune the reflex section of a horn/reflex combination. The reply was that such a combination had not been modeled. It has now. I direct you again to the impedance chart in *Fig. 7*.

After sealing the box, I tuned the cabinet with the duct length that gave the highest SPL at 40Hz without degrading performance at higher frequencies. The result is seen in the solid trace, with an impedance null at 40Hz flanked by peaks above and below it. This is classic ducted-port tuning.

Another impedance null occurs at 125Hz, representing the Fh of the horn (the beginning of the horn passband), followed by a peak at 160Hz that represents the ¼ wavelength resonance of the horn mouth. Nominal woofer impedance then settles in at 9Ω, while the three tweeters paralleled never go below 18Ω. Should you use a driver that differs significantly from an f_S of 55Hz, measure for f_S(h) with the porthole cover off, and then adjust the duct lengths for an Fb that matches it.

Like the Snails before it, the DR10 gives high frequency performance that folded horns are not supposed to deliver. Louis McClure's article refers to the "mass rolloff" of his bass horn, which begins at 390Hz. Not having seen his design, I cannot comment on why his bass horn rolls off at 390Hz, but I can state unequivocally that mass rolloff is a myth. Calculated from a formula derived by Keele ("Low Frequency Horn Design Using Thiele/Small Driver Parameters," AES Reprint #1250, 1977), stating that $Fhm = 2f_S/Q_{ES}$, mass rolloff for the Eminence™ driver in the DR10 should occur at about 407Hz; clearly, it does not.

I certainly would not challenge the credentials of the redoubtable D.B. Keele, but when it comes to denying the existence of mass rolloff, I have cooked the pudding in which the proof does lie. Keele was wrong; the rolloff of high frequencies in a folded horn has no connection with the driver T/S parameters and is purely a result of out-of-phase reflections that occur within the horn. This can be addressed through careful throat design, keeping horn bends to a minimum, and rounding all bends. I plan to experiment further to see whether alternate construction methods can achieve even broader bandwidths and flatter response from folded horns–based on my results with the DR10 I think that the outlook is very promising.

I'm forwarding this article to my potential manufacturer, to see how he next wishes to proceed. Be assured that, whatever the result, you'll see it here on the pages of *audioXpress*. In the meantime, don't wait to buy one of these babies.

Based on what manufacturers are getting for T/S boxes, I'd expect the commercial version to go for at least $800. Head on out to your workshop and build your own (*Table 2*). You won't be disappointed, I guarantee. ❖

Reference MM de CAPO Speaker

Reviewed by Joseph D'Appolito and Ken and Julie Ketler

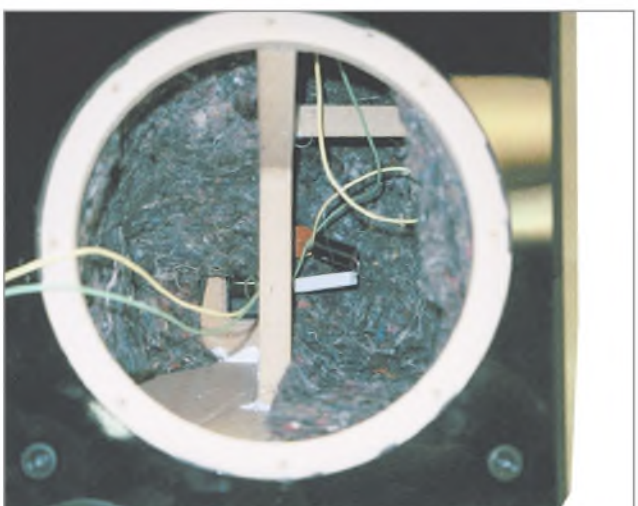
■ **PHOTO 1: MM de CAPO loudspeaker on stand.** ***(Photos by Ken and Julie Ketler).***

■ **PHOTO 2: MM de CAPO rear (left) and front (right).**

Reference 3A, 342 Frederick St., Kitchener, Ontario, N2H 2N9 Canada; 519-749-1565, Fax 519-749-2863, www.reference3a.com.

I ran a series of impedance, frequency response, and distortion tests on the MM de CAPO loudspeaker from Reference 3A. *Figure 1* is a plot of system impedance magnitude. At low frequencies the plot displays the double peaked curve of a vented system. The impedance minimum of 5.24Ω at 45Hz indicates the vented-box tuning frequency.

There is a second local impedance minimum of 5.04Ω at 210Hz. Impedance phase lies between +40° and −47° over the full audio range. Fortunately, these rather large phase angles occur at relatively high impedance values. With minima in the range of 5Ω, Reference 3A's 8Ω rating for this system seems a bit high.



■ **PHOTO 3: Inside the cabinet.**

FREQUENCY RESPONSE

The MM de CAPOs are supplied in mirror-image pairs. The manual states that the speakers are optimally set up with the speakers facing forward (no toe-in). The two speakers and the listening position should be set up to form an equilateral triangle. This implies an optimum listening axis 30° off the on-axis position. Polar responses taken every 10° confirmed that the 30° off-axis position produced the smoothest response.

Figure 2 shows the MM de CAPO's full-range frequency response at the 30° position. This response is obtained as a combination of the far-field quasi-anechoic response and properly summed near-field woofer and port responses. The microphone was placed along the baffle centerline at a distance of 1.2m to produce the far-field response. The near- and far-field responses were then spliced together at 210Hz to produce the full-range response.[1]

The response shown in *Fig. 2* has been normalized to 1m to obtain system sensitivity. Sensitivity averages 86.6dB in the two oc-

ABOUT THE AUTHOR

Joseph D'Appolito, regular contributor and author of many papers on loudspeaker system design, holds four degrees in electrical and systems engineering, including a Ph.D. Previously, he developed acoustic propagation models and advanced sonar signal processing techniques at an analytical services company. He now runs his own consulting firm specializing in audio, acoustics, and loudspeaker system design. A long time audio enthusiast, he now designs loudspeaker systems for several small companies in the US and Europe.

taves between 500Hz and 2kHz. This is substantially less than the 92dB figure quoted on the spec sheet for the MM de CAPO. Response does reach a level of 91dB at 680Hz, but it shelves down to 85dB at 2kHz. Relative to the midband average level, the low frequency −3dB point is 45Hz. There is also broad response peak of about 2dB centered on 90Hz.

The MM de CAPO has two pairs of binding posts for bi-wiring. This allowed me to measure the response of the individual drivers. The result is plotted in *Fig. 3*. Crossover occurs between 2.2 and 3kHz. There is no woofer crossover network. The woofer runs full range. Woofer response simply shelves down by about 15dB above 3kHz. Tweeter response is relatively smooth, but the woofer response shows a great deal of ripple above 4kHz. This ripple also shows up in the overall system response.

CUMULATIVE SPECTRAL DECAY

The MM de CAPO's cumulative spectral decay (CSD) response is presented in *Fig. 4*. This waterfall plot shows the frequency content of the system response following a sharp impulsive input at time zero. On the CSD plot, frequency increases from left to right and time moves forward from the rear. Each slice represents a 0.06ms increment of time.

The total vertical scale covers a dynamic 32dB range. Ideally the response should decay to zero instantaneously. Inertia and stored energy that take a finite amount of time to die away, however, characterize real loudspeakers. A prominent ridge parallel to the time axis indicates the presence of a strong system resonance.

The first time slice in *Fig. 4* (0.00ms) represents the system frequency response. There are two strong ridges in the plot at 4kHz and 11.8kHz. The ridge at 4kHz takes over 2.5ms to decay. Surprisingly, both ridges come from the woofer. Tweeter decay time is quite good.

SYSTEM STEP RESPONSE

The 602's step response is shown in *Fig. 5*. The rise time is somewhat slower than many other speakers I have tested; however, the woofer and tweeter arrive at the same time and rise together. The overall step response is the best I have measured in this series of tests. The excess group delay (plot not shown) is 100μs or less from 300Hz to 20kHz. The MM de CAPO is essentially time-coherent.

POLAR RESPONSE

Polar response is examined in *Figs. 6–10*. *Figure 6* is a waterfall plot of horizontal polar response for the left-side speaker. The curves are plotted in 10° increments from 60° left (−60°) to 60° right (+60°) when facing the speaker. All off-axis plots are referenced to the +30° response, which appears as a straight line. (Remember the +30° position is the preferred listening axis.)

The plotted curves show the change in response as you move away from the 30° line. Smaller positive angles and negative angles move the listener outside of the primary listening area.

The responses above +30° are relatively smooth replicas of the 30° line. This should produce good stereo imaging for listeners between the speakers. However, the curves do degenerate rapidly as one moves beyond the span of the speakers. This will produce wall reflections with a much different spectral balance than the first arrival sound. I suspect the perceived spectral balance will be very room dependent. Responses at 0, +30 and +60° are plotted in *Fig. 7*. The 30° curve is seen to be the smoothest response.

Without building a special jig to obtain the required compound angle, I could not measure response changes in the vertical plane containing the 30° off-axis position. For this reason, vertical responses were measured in the on-axis plane. The changes observed should be similar to those seen at the 30° position.

Figure 8 is a waterfall plot of vertical polar response. Responses are shown in 5° increments from 20° below (−20°) the tweeter axis

TECHNICAL DATA

Type of speaker:	Medium-size monitor
Music peak power:	120W peak
Music impulse power:	80W
Continuous power rating:	60W
Frequency response:	44–20kHz ±3dB
Efficiency:	92dB 1W/m
Recommended amplifier:	5–100W
Crossover frequency:	3.000Hz (tweeter only)
Impedance:	8Ω nominal
Woofer: (no crossover)	21cm, direct coupled
Tweeter:	26mm latex-coated silk dome with closed back chamber
Weight net:	20kg/43 lbs
Dimensions:	38cm/15″ high, 28cm/11″ wide, 33cm /13″ deep at the base
Finish:	Gloss black

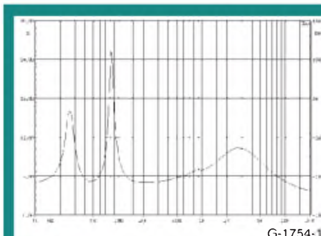


FIGURE 1: MM de CAPO system impedance.

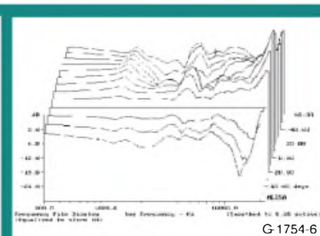


FIGURE 6: Horizontal polar response.

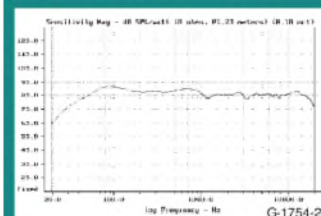


FIGURE 2: Full-range frequency response of the MM de CAPO (30° right).

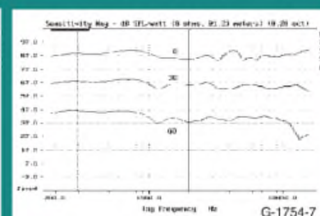


FIGURE 7: Response at 0, 30, and 60° right.

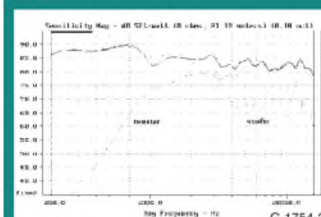


FIGURE 3: MM de CAPO system and driver responses.

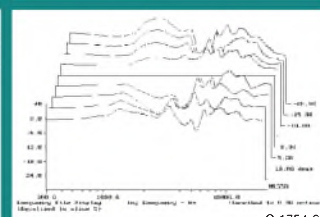


FIGURE 8: Vertical polar response.

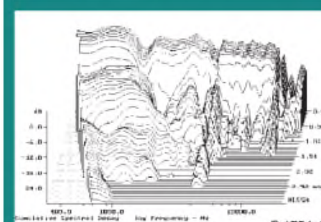


FIGURE 4: Cumulative spectral decay.

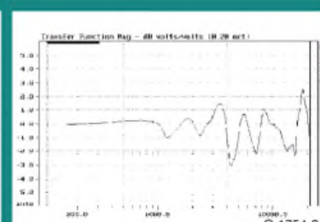


FIGURE 9: Effect of grille on MM de CAPO response.

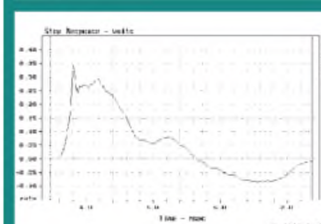


FIGURE 5: Step response.

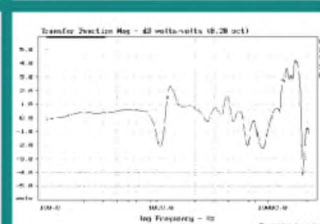


FIGURE 10: Loudspeaker matching.

Reviewed by Ken and Julie Ketler

Located in Ontario, Canada, Reference 3A claims over 30 years of loudspeaker design and innovation. Their current product line consists of five models of loudspeakers, each demonstrating a propensity toward the high end. Although mentioning "high end" equipment may fend off some budget-minded music lovers, Reference 3A has produced a "moderately priced" loudspeaker—the MM de CAPO, which deserves consideration from anyone who is passionate about music listening.

The MM de CAPO is a bi-wireable two-way rear-vented system, which utilizes an 8″ (203mm) woven carbon woofer, a 1″ (26mm) soft textile dome tweeter, and no crossover! Well, almost. This is really a low-calorie crossover—not quite fat free!

Although the woofer is designed to require no electronic filtering to achieve its rolloff slope and frequency, the tweeter has an impedance-compensated first-order circuit. Reference 3A claims that this configuration provides a frequency response of 44Hz–20kHz and a smooth phase response that is constant within ±10°. The rated sensitivity is a healthy 92dB (1W/1m), which makes this a very good candidate for those readers with lower-powered amplifiers. Quality versus quantity, don'tcha' know?

FIRST IMPRESSIONS

This speaker has a footprint of 15″ (38cm) (H) × 11″ (28cm) (W) × 13″ (33cm) (D). Most reasonable human beings might be somewhat hard-pressed to call this a "mini-monitor," however, it's a nice size and should be easy to integrate with any modern decor. The first thing that a person with an eye for furniture will notice about the enclosure is its mirror-smooth glossy black finish. Very attractive indeed. The front face is sloped in order to time-adjust the two drivers...and it looks pretty cool, too!

Since the MM de CAPO weighs 60 lbs (27Kg) per pair (we weighed them), it's not difficult to see why the deliveryman was cursing our name as he walked down our steep driveway and hopped back into his truck. Such a heavy enclosure may be bad for the back, but is likely to be good for music. Such thick walls will tend not to vibrate and impose a sonic signature of their own. As a result, the woofer is allowed to do its own job.

Ah, yes, the woofer—the first noticeable thing was that the phase plug in the center is anchored to the driver frame/magnet and not the cone, i.e., it does not move. The fibrous texture of the woven carbon material appears to be a great scheme for damping driver resonances and breakup modes.

The inside of the cabinet is built like a house! It has plenty of bracing and is just plain solid. Reference 3A's choice of internal wiring is a bit surprising. They use what appears to be just flopsy #18 AWG hook-up wire, not quite what you might expect from a high-end company.

Also, the enclosure has a generous amount of damping material, but when we gazed in, we noticed it was loosely covering the bass reflex vent of one of the loudspeakers.

COMPANY SETUP RECOMMENDATIONS

The user guide accompanying the MM de CAPO loudspeakers is a mere four pages of text, but it includes a great deal of information regarding room positioning/acoustics, height, cables, and amplifiers. Reference 3A suggests putting the MM de CAPO on stands approximately 26″ high and forming an equilateral triangle with the listening position. They do not recommend toeing-in and stress that "early reflections will collapse the soundstage." Reference 3A also mentions that the MM de CAPO doesn't "lack bass performance when placed a meter or two out from the walls."

OUR SETUP

We mounted the MM de CAPOs just as Reference 3A suggested with respect to our listening position. We drove them with both tube (Valve Audio Laboratory VAA-100, 30W) and solid-state (AudioSource Amp Two, 80W) amplifiers on separate occasions and compared them to many different pairs of custom-built speakers. During casual listening sessions, we played many of our own personal CDs, but for level matching and critical evaluation, we used the *Hi-Fi News & Record Review's* CD Test Disk III. Here's what we found:

EVALUATION

TEST 1
TRACK 2—Jerusalem/Parry

JK: The MM de CAPO loudspeakers have a very smooth and full stereo sound in this piece. All the instruments sound bright and clear.

KK: Excellent front-to-back depth, the Philharmonia Chorus sounds very large.

TEST 2
TRACK 4—Trumpet Concerto in C/Vivaldi

JK: I am able to place where the trumpets are playing in this most beautiful piece (simply my favorite, maybe that makes me biased, nah). When the entire orchestra joins in with the trumpets, the room is filled with a very rich and full sound as if I am witnessing the show live.

KK: The string sections have a nice three-dimensional "silky" quality. But I must disagree with my beautiful wife, though. To my ears, the stereo placement of the two trumpets in this piece is slightly smeared amid the wide soundstage (she's probably just biased).

TEST 3
TRACKS 5/6—Peter & the Wolf (narr)/Prokofiev

JK: During the clarinet solo, the instruments sound as though they are coming from the back of the speakers instead of from the front. I enjoy predictability, so this "hidden" sound bothered me a bit. Yet, I know many people opt for this effect.

The tympani and drum ensemble sounds loud and clear, filling the room with a rich, fat sound. As I listen very critically, I do find that the tympani sounds a little "echo-like" as if it's being "filtered." Even though this piece does have a full stereo sound all together, the stereo placement is difficult to hear.

KK: The sibilance of Sir John Gielgud's voice is very smooth and intimate. The oboe, clarinet, and bassoon all have a realistic breathy quality. The horns, however, lack the brassy "zing" that is present in this recording. The tympani and bass drum sound very deep and dynamic, but the attack of the mallets hitting the drum skins is veiled due to a somewhat boomy bass overhang. The tambourine during the "procession" leaps forward right out of the speakers!

TEST 4
TRACK 7—Welcome, Welcome/Purcell

JK: The violins appear to be playing on the left and the singers are rich-sounding in the middle. The overall sound is very full, filling the space in front of me in an arch shape.

KK: The vocals sound very nice in this piece and seem to come from behind the speakers. The smooth drone of the chamber organ is deep and encompassing.

TEST 5
TRACK 10—Corkhill (piece 2)

JK: The drums sound lower-pitched than they do with our speakers; both sound very different as if the drums were playing in different octaves. This piece proves to be a very interesting comparison with very contrasting results. Similarly though, in this piece, both speakers have a great stereophonic effect.

KK: Fourteen seconds into this all-percussion piece, there erupts a burst of snare and bass drums that scares the whoopee out of me every time! Very dynamic. Again, however, the impact of the bass drum (left channel) is somewhat softened and the snares are missing some "bite."

TEST 6
TRACK 14—Rio Napo RSS Demo

JK: This funky piece has a full 3-D sound with the MM de CAPO speakers. The music circles the room with a pleasant soft sound, which is very enjoyable to listen to.

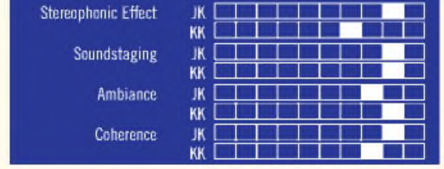
KK: The de CAPO really shines on this track. Its 3-D soundstaging is absolutely super! The somewhat chubby bass quality of this loudspeaker sounds perfect with this pop-style tune.

FINAL THOUGHTS

Overall, the MM de CAPO provides a very wide soundstage that appears to originate from far outside the confines of the enclosure. Its response is smooth and unfatiguing while providing great detail in the upper high frequencies. Low frequencies are definitely prominent, however, somewhat undefined and occasionally boomy.

On various rock and jazz tracks, the MM de CAPO lacks "thwack," most notably on snare drums. On rapid bass progressions, it is occasionally difficult to distinguish individual notes. With more mellow jazz and various classical pieces, the MM de CAPO sounds full-bodied and clear with no noticeable flaws.

For listeners who enjoy a larger-than-life soundstage and are currently set up with an active crossover and subwoofer, the MM de CAPO will provide you a potentially wonderful listening setup. Those of you who own low-powered amplifier configurations (single-ended triodes, for example) will be pleasantly surprised with the sensitivity and fullness of this comparatively compact system.



ABOUT THE AUTHORS

Julie and Ken Ketler are the proud parents of an eleven-month-old boy, Thomas Anthony. Julie is a first-grade teacher who is on a long hiatus to be a full-time mom, and Ken is a technical writer for a chemical lab equipment company in Mass. Thomas enjoys listening to music with his parents and occasionally even sings along.

to 20° above it. Response changes very rapidly as one moves above or below the horizontal plane. This is a direct consequence of the broad driver frequency overlap.

HARMONIC DISTORTION

Harmonic distortion tests were run at an average level of 90dB SPL. Ideally, harmonic distortion tests should be run in an anechoic environment. In practice, it is important to minimize reflections at the microphone during these tests. Out-of-phase reflections can produce false readings by reducing the level of the fundamental while boosting the amplitude of a harmonic. In order to reduce the impact of reflections, I placed the microphone at 0.5m from the loudspeaker.

Second-harmonic distortion was below 1% over most of the audible frequency range. Below 100Hz second-harmonic distortion did rise to 3.6%, but this is still a low figure. Third-harmonic distortion was 1% or less over the full audio range. This is an excellent result.

INTERMODULATION DISTORTION

I next measured intermodulation distortion. In this test two frequencies are input to the speaker. Intermodulation distortion produces output frequencies that are not harmonically related to the input. These frequencies are much more audible and annoying than harmonic distortion.

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

I first examined woofer intermods by inputting 900Hz and 1kHz signals at equal levels. These frequencies should appear predominantly in the woofer output. I adjusted total SPL with the two signals to 86dB at 1m. Because steady tones are used in the IM test, I believed it safer to use a lower power level to prevent possible tweeter damage.

Principal woofer IM products occurred at 800, 1100, 1900, 2800, and 2900Hz. Total IMD was 1.1%. This figure is one of the highest measured in this series of tests. Much of this distortion arises in the tweeter due to the slow rolloff of its 6dB/octave crossover that lets too much low frequency energy into the tweeter.

I measured tweeter intermods with a 10kHz and 11kHz input pair also adjusted to produce 86dB SPL at 1m. I observed IM products at 8, 9, and 12kHz. Total distortion was 0.2%. When frequencies are limited to those the tweeter is designed to handle, tweeter distortion performance is quite good.

The last IM test examines cross-intermodulation distortion between the woofer and tweeter using frequencies of 900Hz and 10kHz. (A 1kHz signal would produce intermods that fall on harmonic distortion lines, confusing the results.) In typical systems with woofer and tweeter crossovers, the crossovers should prevent high-frequency energy from entering the woofer and low-frequency energy from entering the tweeter. In the case of the MM de CAPO, IM products appeared at 8.2, 9.1, and 10.9kHz at a level of 0.2%. In absolute terms this may be an acceptable figure, but the majority of the systems I have tested typically produce values in the range of 0.05 to 0.07%.

ADDITIONAL TESTS

I conducted all of the above tests with the grille off. *Figure 9* shows the MM de CAPO's system response with the grille on relative to the response with the grille off. That is, it plots the change in response under the two conditions. Below 1kHz the grille has little effect. Above 1kHz, however, the grille causes ragged response deviations of +2.5 to −3dB. As usual, the grille has only cosmetic value.

Two samples of the MM de CAPO system were available for testing. All of the tests described so far were conducted on one sample. Frequency response matching of the pair is shown in *Fig. 10*. This is a plot of the response difference between the first and second samples. Between 1 and 10kHz the two speakers match within 2dB. Above 10kHz matching degrades to ±4dB.

CLOSING COMMENT

The MM de CAPO is the first speaker I have tested that is essentially time-coherent, but this attribute comes at a price, namely, poor polar response and somewhat higher distortion. I leave the audible consequences of these results to the reviewer. ❖

Manufacturer's Response:

Reference 3A (initially known as 3A of France) has been in the forefront of innovative loudspeaker design for over 30 years. Some of its complex designs utilized concepts and ideas not well known 20-30 years ago. A 3-way design with motional feedback technology called Master Control is still used in many homes and studios around the world, even after 26 years. Also from the same era is the Triphonic, a self-powered coffee table sub-satellite system, one of the first of its kind.

A few years ago, Reference 3A adopted a simpler approach to enhancing loudspeaker performance. By removing the crossover from the main driver, the company addressed the potential damage complex crossover networks imposed on the flow of the musical signal. This required a new, hand-built driver tuned to cover a wide range. The other approach was to emphasize the actual listening process and not rely solely on measurements. Necessary adjustments to the design would be made mainly according to listening tests.

Reference 3A MM de Capo is the latest model following this approach. It is tuned by ear to elevate the enjoyment of music. With coherent and balanced presentation, these units are crafted to allow the listener to enjoy music for much longer periods without the listening fatigue. The drivers are still hand-built by the designer and founder of Reference 3A, Daniel Dehay, in Europe.

As for Julie and Ken Kelter's subjective listening tests, I like Julie's (perhaps biased) comment, describing a crescendo from Vivaldi's Trumpet Concerto in C: "When the entire orchestra joins in with the trumpets, the room is filled with a very rich and full sound as if I am witnessing the show live."

I would like to clarify one small point regarding the MM de Capo's internal wiring. The wires are actually the best and most expensive 16 AWG wire van den Hul, in the Netherlands, produces. They are MCS-16 matched crystal silver conductors with high-grade Teflon® insulation.

A note on testing: The MM de CAPO was tested in the laboratories of Audio and Acoustics, Ltd. using the MLSSA and CLIO PC-based acoustic data acquisition and analysis systems. Acoustic data was measured with an ACO 7012 ½″ laboratory-grade condenser microphone and a custom-designed wideband, low-noise preamp. Polar response tests were performed with a computer-controlled OUTLINE turntable on loan from the Old Colony division of Audio Amateur Corporation.

REFERENCES

1. J. D'Appolito, *Testing Loudspeakers*, Audio Amateur Corporation, Peterborough, NH, 1998.

Manley Stingray Amp

Reviewed by Charles Hansen and Nancy and Duncan MacArthur

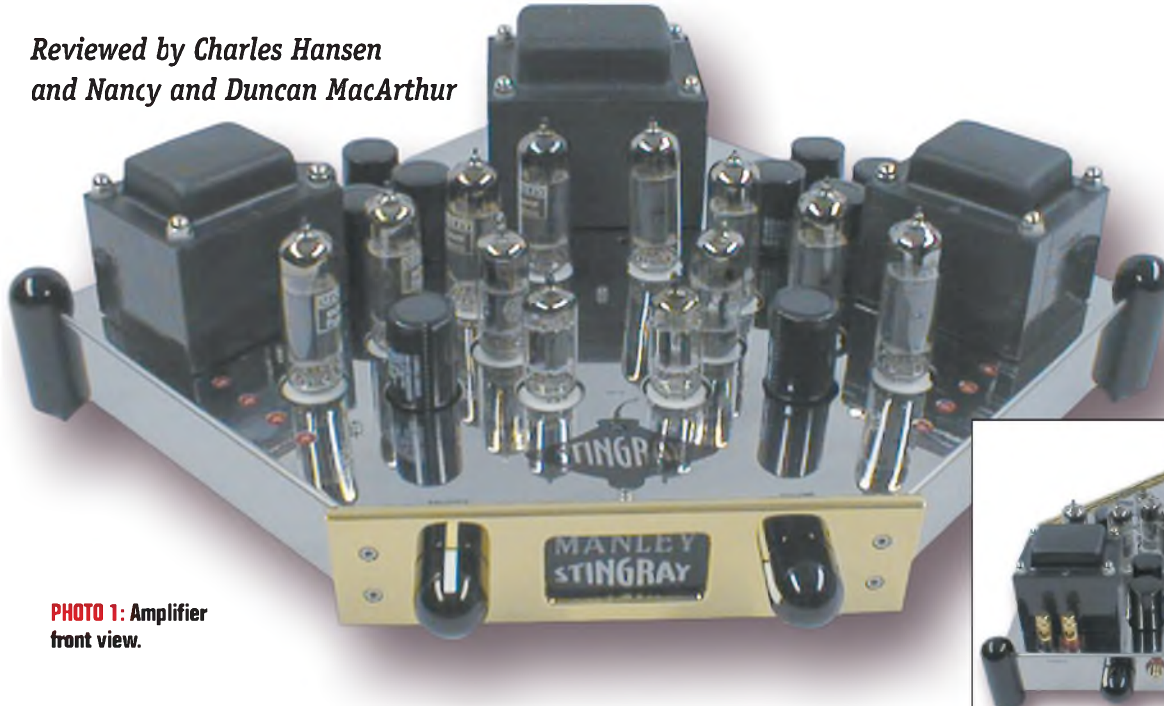


PHOTO 1: Amplifier front view.

PHOTO 2: Amplifier right/rear view.

Manley Laboratories, 13880 Magnolia Ave., Chino, CA 91710, (909) 627-4256, www.manleylabs.com, emanley@manleylabs.com. Assembled: $2250. Dimensions: 6.5″ H × 19″ W × 15″ D; net weight: 30 lbs. Limited five-year warranty (tubes six months).

The Manley Laboratories Stingray is a push-pull stereo integrated amplifier rated for 50W per channel. The heavy-gauge, high-polish stainless steel amplifier chassis (designed and signed by EveAnna Manley) is a square whose corners have been truncated front and rear—more so in the front. Pointed solid aluminum footers are located on each side of the chassis, and two more elastomer footers are mounted underneath, for a four-point stance. *Photo 1* shows the gold-plated front panel, which has the balance and volume controls, and a power-on indicator lamp behind the logo.

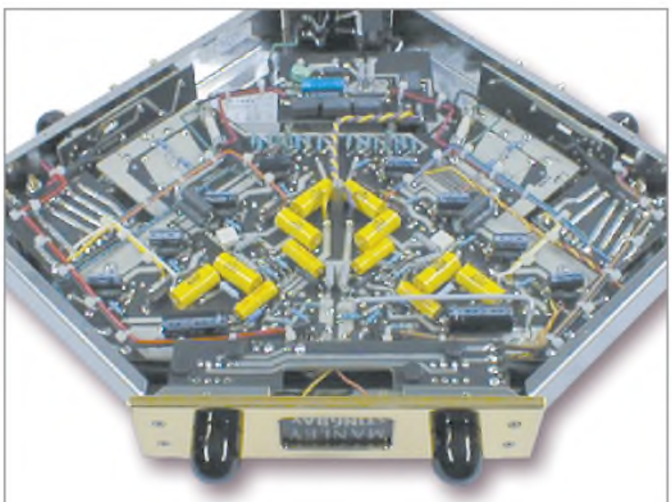


PHOTO 3: Amplifier interior view.

CONSTRUCTION

The right rear view (*Photo 2*) shows the four-position input selector switch and four high-quality gold-plated Teflon™-insulated RCA input jacks, and high-quality gold-plated ⅜″ hex-nut speaker binding posts. (RCA jacks and binding post are Manley's own design and manufacture.) The amplifier design is optimized for 5Ω speakers. To the right is a short rear panel with the IEC power receptacle and power switch.

On the top/front surfaces, eight test-point jacks allow you to monitor the bias for the eight EL84 output tubes. A common bias-ground jack is located front/center at the tail of the silkscreened Stingray. The bias for any of the EL84 output tubes (250mV DC optimum) can be individually adjusted using trimpots recessed near the front of the power transformer.

The unit is furnished with a heavy power cord. The line fuse is located in a drawer in the IEC power receptacle. Two B+ fuses are located under the bottom cover. The front-panel indicator lamp also resembles a 3AG fuse. The third pin of the AC receptacle is connected to the chassis. There is adequate finger space under the unit to easily lift it.

Photo 3 shows the amplifier with the perforated bottom cover removed. The majority of the circuitry is contained on one large PC

ABOUT THE AUTHOR

Charles Hansen has worked as an engineer since 1967, and has five patents in his field of electrical engineering. He began building vacuum-tube audio equipment in college. He plays jazz guitar and enjoys modifying guitar amplifiers and effects to reduce noise and distortion, as well as restoring audio test equipment.

board, signed by the constructor. Three small PC boards above each of the transformers provide a means of connection and, in the case of the Manley-designed power transformer, are fitted with four solid-state 1000V 2A rectifier bridges and additional passive filter components. The 12AT7 and 6414 heaters receive rectified 12V DC.

Three additional PC boards carry the front volume/balance pots and, at the rear/sides, the input jacks and selector switches. The few areas of interconnect wiring are beautifully dressed, with twisted pairs and coax used where needed.

The components are all of high quality, with Cornell-Dubilier, NIC, and Shoei electrolytics; custom polypropylene rolled film and foil caps; and military-style 1% metal film and 1% to 5% wire-wound resistors.

The illustrated owner's manual is complete and entertaining, with an excellent troubleshooting section. There are additional sections devoted to Optimizing Your Sound System, Replacing the Tubes, and Setting the Bias.

The Stingray is shipped with all tubes installed, held securely by a form-fitted foam rubber block with a hole for each tube. Very practical!

TUBE-POLOGY

A schematic was not supplied with the Stingray, but is available to registered owners upon request. The input selectors and the Noble conductive plastic balance and volume controls form a passive stereo line stage. Inputs are labeled CD, Video, Tuner, and Aux.

The input stage for each channel is a 12AT7 dual-triode, followed by a 6414 dual-triode phase splitter. The plates of the 6414 each drive a pair of parallel-connected EL84s in ultralinear push-pull. The output tubes receive grid bias from a negative bias supply. The output transformer secondaries (optimized for 5Ω) can be connected to speakers with impedances of 3Ω to 10Ω. A very low level of global feedback is used in the Stingray amplifier. The manual cautions against operating the amplifier without a speaker load (open circuit).

There are ten Cornell-Dubilier type 380LX high-voltage electrolytic filter caps protruding through holes in the top of the chassis. Four series-parallel connected 330µF 350V DC caps in each channel provide the main B+ filtering, with a 220µF 450V DC cap next in the pi filtering string.

Manley supplied the following tube complement with the Stingray:

2—"Manley" 12AT7WAs from EI
2—GE NOS 6414s
8—Russian "Manley" EL84Ms (6π14 Military EL84/6BQ5 tube)

The two Manley-designed output transformers do not appear as large as those of some other power amplifiers of this rating, but this is deceiving since the core stack is fairly high with one end-plate under the chassis.

PRELIMINARY CHECKOUT

There was no noise during power-up, and only a slight noise during shutdown. There was no hiss and a very low-level hum from both channels with my ear against the speaker. This was independent of the volume-control setting.

TABLE 1
MEASURED PERFORMANCE

PARAMETER	MANUFACTURER'S RATING	MEASURED RESULTS
Gain	37dB, maximum volume	38.6dB, 8Ω load
Sensitivity load	185mV for 50W into 5Ω	233mV, 50W 8Ω
Signal to noise	87dB A-WGT	
Frequency response	15Hz–40kHz, –1dB	15Hz–42kHz, –1dB
Total harmonic distortion	0.25% 5W, 1kHz (–55dB)	0.26% 5W, 1kHz 8Ω
Volume control tracking	0.25dB L/R	0.2dB max at 11 o'clock
Input impedance	50k nominal	33k8 to 44k8
Power output (RMS)	50Wpc 1.5% THD, 1kHz, 5Ω	50Wpc 3% THD 1kHz 8Ω (45Wpc 1.5% THD)
IMD—CCIF (19 + 20kHz)	N/S	0.051% at 12V p-p out
MIM (9 + 10.05 + 20kHz)		0.043% at 12V p-p out
Output impedance	Load Z	3.46Ω 1kHz
Power consumption	200W min, 370W max	
Damping factor	10	

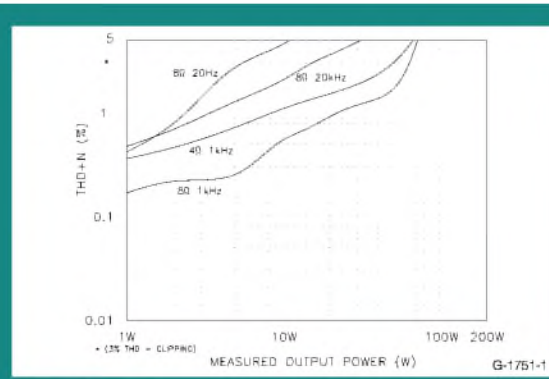


FIGURE 1: THD+N versus output power.

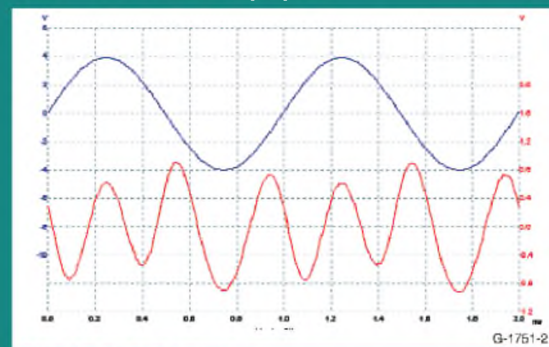


FIGURE 2: Residual distortion.

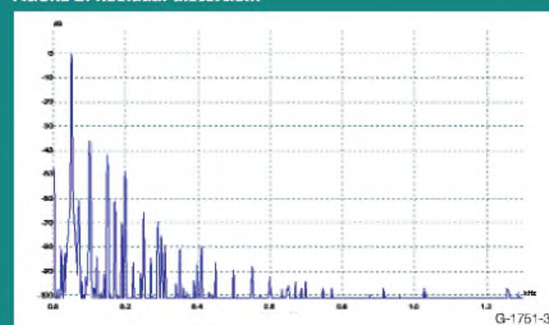


FIGURE 3: Spectrum of 50Hz sine wave.

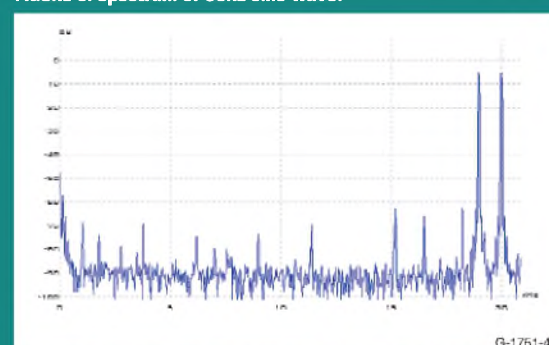


FIGURE 4: Spectrum of 19kHz + 20kHz intermodulation signal.

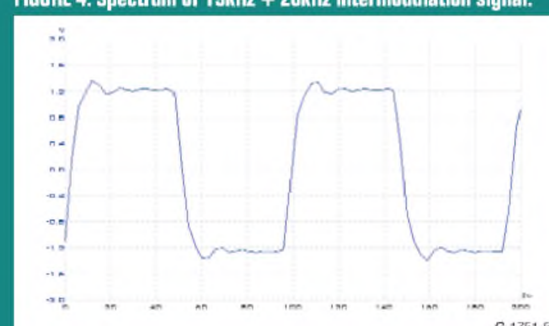


FIGURE 5: 10kHz square-wave response.

MEASUREMENTS

I operated the Stingray at 10W into 8Ω for one hour. The bias current after this run-in was 240mV in each of the eight output tubes, which is certainly close enough to the specified 250mV. I made all measurements with the level control at 12 o'clock and an 8Ω load, unless otherwise noted. The overall gain at this setting was 24dB. The distortion was essentially the

CRITIQUE

Reviewed by Nancy and Duncan MacArthur

The Manley Stingray arrived in a single, light (30 lbs—no single-ended output transformers here) box. The amplifier and tubes, which were already mounted, were well-protected by custom-made foam inserts inside the box. The shipping materials were in much better shape than those of heavier amplifiers we have reviewed.

AESTHETICS

The Stingray immediately caused dissension in our household. Duncan and I were entranced by its looks. "Retro," I breathed. "Beautiful," said Duncan. Colin, our ten-year-old, raised his eyebrows. "Ugliest thing I've ever seen," he said.

Regardless of your opinion of its looks, you probably won't find a CD player that "matches" it. On a more practical note, Manley suggests (and we agree completely) that the Stingray may not fit in a standard enclosed equipment rack.

We think the Stingray looks good enough that you won't wish to hide it anyway. But this is a tube amplifier that generates a normal amount of heat. As all the tubes are completely exposed, the usual cautions about children and pets apply. The Stingray is supported by two sharp and two rubber pointed feet. We placed pennies under these points, but the machined "footer-holders" that are widely available (for somewhat more than a penny) would work as well or better.

Given the price, the "fit and finish" of the entire amplifier is excellent. The small front panel holds volume and balance controls that operate smoothly and flawlessly in our sample. The placement of input selector switches on the rear panels would be inconvenient if frequent source changes were likely. As we only use one source (CD player), this characteristic didn't bother us. Similarly, the Stingray's power switch is located on the rear panel.

The location of these switches may be more than an inconvenience, as the speaker terminals are located on top of the chassis near the selector switches. Like most tube amplifiers, the Stingray should not be operated without a speaker load. Frequent reaching behind the amplifier to operate the input selectors could cause the speaker connections to loosen accidentally with potentially disastrous results.

LISTENING EXPERIENCE

All our auditioning was done using Genesis 400 speakers. These three-way floor-standing systems (not bi-amplified) are rated at 89dB/W efficiency but are 4Ω designs with a fairly difficult impedance curve. We directly compared the Stingray with a pair of VTL "Tiny Triode" amplifiers. The VTL amplifiers use the same output tube complement (four EL-84s per channel) but strap these tubes for triode operation; as a result the output power of the VTLs is only 25W per channel. For reference purposes, the VTL amplifiers would receive sixes in all four categories in the ratings box.

The Stingray changed in small but important ways during the break-in period. Right out of the box, the bass and dynamics were impressive; however, the treble was a bit bright and the Stingray sounded more like a very good "hi-fi" amplifier and less like real music. About 50 hours of break-in was required before serious listening.

Following break-in we listened to the *Hi-Fi News and Record Review* disk III (track 2: Parry's "Jerusalem," track 4: Vivaldi's trumpet concerto, tracks 5 and 6: excerpts from Prokofiev's "Peter and the Wolf," track 7: Purcell's "Welcome, Welcome Glorious Morn," track 10: a Corkhill percussion piece, and track 14: Rio Napo RSS demo). We also played favorite tracks from a wide variety of musical genres ranging from classical to rock 'n' roll. In all cases the Stingray was turned on at least 30 minutes before serious listening.

This amplifier's ambiance recovery is impressive. The Stingray clearly defined the open-air space around the choir in "Jerusalem" and the acoustic space around the trumpets in the Vivaldi. In "Misguided Angel" on the Cowboy Junkies' *The Trinity Session* (BMG Music Canada Inc. compact disk, 8568-2-R), the acoustic volume of the church in which the song was recorded stood out clearly.

The Stingray's lateral imaging is quite good. Its placement of trumpets in the Vivaldi concerto was precise and stable, as was its placement of the choir and trumpets in "Jerusalem." It gave an astonishing width of sound, extending well beyond the speakers, to the Rio Napo demonstration.

For an unusual imaging experience, try playing The Charlie Daniels Band's "The Devil Went Down to Georgia" (*Fiddle Fire*, Blue Hat Records compact disk, BLH-9703-2) on this amplifier and listening for the screech shooting from left to right across the soundstage partway through the song. The image depth extended back only a short distance behind the speakers on most tracks and was, in our opinion, the Stingray's weakest characteristic.

This amplifier produces a strong, natural bass. The bassoon on tracks 5 and 6, "Peter and the Wolf," sounded like a real bassoon playing in the listening room. The sound of the timpani was deep, well-defined, and almost scary.

Since our Genesis speakers are "only" flat to 40Hz, we were unable to check out the Stingray's capacity to reproduce extremely deep bass. We tried playing Tchaikovsky's "1812" Overture (Cincinnati Symphony Orchestra, Erich Kunzel, Telarc compact disk, CD-80041). The cannon shots sounded surprisingly realistic, although, of course, they lacked the last octave or more of bass.

The Stingray's midrange is excellent. One of this amplifier's greatest strengths is its superb reproduction of the human voice. Whether we were listening to opera or Bob Dylan, every note, rasp, and tremor stood out clearly. Solo piano pieces also sounded completely realistic.

As illustrated in the Rio Napo demo, the Stingray's highs are very clean and clear. In "Peter and the Wolf," the flute sounded like a real flute. One caution: if your speakers tend towards brightness, a home audition is definitely in order before you purchase the Stingray. The Stingray was quite merciless in reproducing poor recordings.

The Stingray provides exceptionally fast transient response. The attack on the drum sounds in the Corkhill piece was fast and dynamic. This amplifier also handles extreme differences in loudness gracefully. On "Peter and the Wolf," the dynamic difference between single groups of instruments and the entire orchestra was staggering, as was the dynamic contrast between the soft and loud drums on the Corkhill piece. The Stingray never sounded compressed, even at earsplitting levels.

This amplifier consistently delivers a realistic, natural sound. In "Peter and the Wolf," Purcell's "Welcome, Welcome, Glorious Morn," and the Rio Napo demonstration, the individual instruments were easily distinguishable and well-defined. The massed strings and trumpets in track 4, the Vivaldi trumpet concerto, sounded realistic and never got "steely." As we mentioned earlier, voices and piano were amazingly detailed and realistic.

A final note: we seldom listen to "sound effects" recordings, but track 18 of the test disk included a recording of a dentist's drill that the Stingray reproduced so faithfully at realistic volumes that it always cleared the room.

FINAL THOUGHT

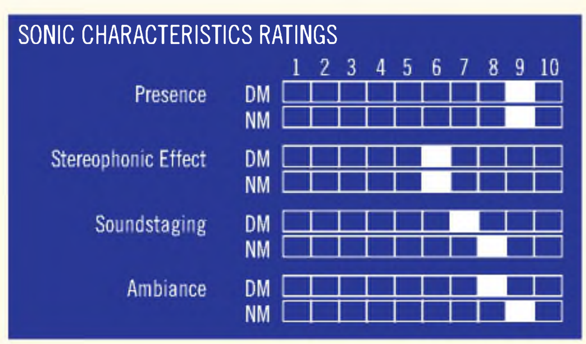
NM: The nines in the ratings box do not convey the sheer pleasure of listening to this amplifier. Very highly recommended, unless depth is your main purchasing criterion or you own bright speakers.

DM: *Is the Stingray the best integrated amplifier available?* I don't know—I haven't heard all of the contenders.

Is the Stingray perfect? No. The imaging, especially the reproduction of depth, was very good but not "jaw dropping." The highs were crystal clear and could become excessive if the Stingray were coupled to bright speakers.

Is the Stingray the best amplifier I've ever had in my listening room? Yes.

Would I consider buying the Stingray? Yes; and so should you if your budget allows.



ABOUT THE AUTHORS:

Nancy and Duncan MacArthur live in northern New Mexico with their young son, Colin. Nancy has written for *Audio Amateur* and *Speaker Builder*, as well as for *Glass Audio*. Duncan works as a physicist. In addition to equipment reviewing, he designs and constructs audio components and state-of-the-art detection systems.

same for each channel. The right-channel test data is presented here, and summarized in *Table 1*.

The Stingray does not invert polarity. Input impedance for both channels measured 44k8 at minimum volume, and 33k8 at the maximum volume setting. The full-volume gain at 2.83V RMS output into the 8Ω load was 38.6dB. The output impedance was 3.5Ω at 1kHz, and 3.7Ω at 20kHz. The amplifier frequency response will be affected by any speaker impedance variations.

The frequency response for the Stingray was within ±3dB from 9Hz to 55kHz, with 0dBr defined as 2.83V RMS at 1kHz into 8Ω. It was –0.7dB at 20Hz and –0.3dB at 20kHz. The HF gain rolled off smoothly to –8.3dB at 80kHz, then there was a gain "peak" to –5dB at 97kHz. When I connected a load of 8Ω paralleled with a 2µF cap, the HF peaking shifted down to 50kHz.

Channel separation measured –70dB at 1kHz, –60dB at 10kHz, and –53dB at 20kHz L-R. *Figure 1* shows THD+N versus output power into 8Ω at 20Hz, 8Ω at 20kHz, 4Ω at 1kHz, and 8Ω at 1kHz (top to bottom at 10W). There was absolutely no strain right up to the point of maximum power.

The clipping level is normally defined as that power where THD+N reaches 1%. However, the baseline distortion in tube amplifiers is fairly high at low power levels, so the generally accepted practice is to use 3% THD+N as the clipping point.

Using a 1kHz signal driving both channels, clipping occurred at exactly 50W into both 4Ω and 8Ω. The power available at 20Hz and 3% THD+N was only 5.5W. At 20kHz I measured 15W into 8Ω at 3% THD+N. The maximum power available at 1kHz was 86W into 4Ω and 78W into 8Ω, at a high 15% THD. The clipping was symmetrical and quite flat-topped for a tube amplifier (56V p-p).

Perhaps the designers caused clipping to occur in the input stage rather than the output tubes. Maximum power at 20Hz was just 21W at 10% THD+N, but at 20kHz I measured 55W at 10% THD+N. My test bench 60Hz mains voltage was 119V AC. After these tests, the power transformer and output transformers were fairly hot.

The distortion waveform for 10W into 8Ω at 1kHz is shown in *Fig. 2*. 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 shows only low-order harmonics, with no evidence of any noise or fuzz.

The spectrum of a 50Hz sine wave at 10W into 8Ω is shown in *Fig. 3*, from zero to 1.3kHz. The THD+N measured 1.84%, with the second, third, fourth, and fifth measuring –36dB, –42dB, –49dB, and –66dB, respectively. There are several artifacts of power supply rectification at 60Hz (–60dB), 120Hz (–84dB), and 180Hz and 240Hz below the –60dB line.

Figure 4 shows the Stingray's output spectrum reproducing a 12V p-p combined 19kHz + 20kHz intermodulation distortion (IMD) signal into 8Ω. The 1kHz IMD product is 0.051%. Repeating the test with a multi-tone IMD signal (9kHz + 10.05kHz + 20kHz, not shown) resulted in a 1kHz product of 0.043%.

The 2.5V p-p square wave into 8Ω at 40Hz showed a tilt of 45°. The 10kHz square wave (*Fig. 5*) contained ringing at the 97kHz peaking measured in the frequency-response test. This ringing was also noticeable at the leading edge of the 1kHz square wave. When I connected 2µF in parallel with the 8Ω load, the leading edge became more rounded, and the ringing frequency dropped to about 50kHz.❖

Manufacturer's response:

Thank you all so very much for the hard work and effort in reviewing our Stingray. We are very pleased it did so well in your stringent battery of tests even at less-than-its-optimized 8 and 4Ω. We've always been a 5Ω company for some reason.

EveAnna Manley
President
Manley Laboratories, Inc.

Tilting With Tubes

In response to a reader request, this author adapted a previously published tilt control design for tubes. **By Scott K. Reynolds**

In a letter published in the 6/99 issue of *Glass Audio*[1], Aaron Freed expressed interest in a vacuum tube version of Reg Williamson's tilt control. I recently built a prototype of such a circuit, and I thought it might be of general interest to *Glass Audio* readers, including Mr. Freed.

FILTER CIRCUIT

Figure 1 shows the IC op-amp version of Mr. Williamson's circuit, which is taken from Fig. 7 of Williamson and Watling's article on their "Back to the Future" preamp.[2] This circuit allows you to tilt the audio frequency spectrum up or down with variable slope around a fixed central frequency, which is about 950Hz in this case. With the potentiometer in its center position, the frequency response is flat.

Rotated in one direction, low frequencies are boosted, and high frequencies are attenuated by the same amount. Rotated in the opposite direction, high frequencies are boosted and low frequencies are attenuated. The maximum action of the control is limited to 6dB of boost or attenuation. The control can be taken out of the signal path by placing SW1 in the flat position, in which case the circuit has unity gain for all audio frequencies.

This circuit is a member of a general class of filters known as infinite-gain single-amplifier filters.[3] It is also rather ingenious because it combines the functions of conventional shelving bass and treble controls in a single control with a minimum number of components. Don't be fooled by the simplicity of *Fig. 1*; this is quite a sophisticated tone control. If the tilt potentiometer has a calibrated scale, it is possible to control the amount of tilt down to a few tenths of a dB.

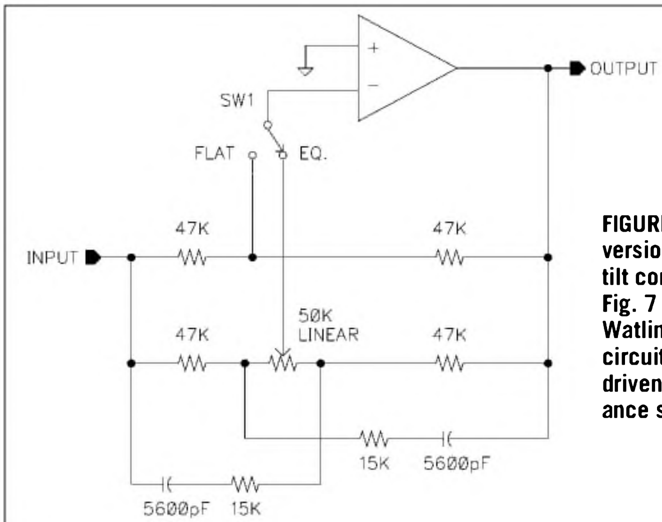


FIGURE 1: IC op-amp version of the Williamson tilt control, taken from Fig. 7 of the Williamson/ Watling article.[1] The circuit is assumed to be driven from a low impedance source.

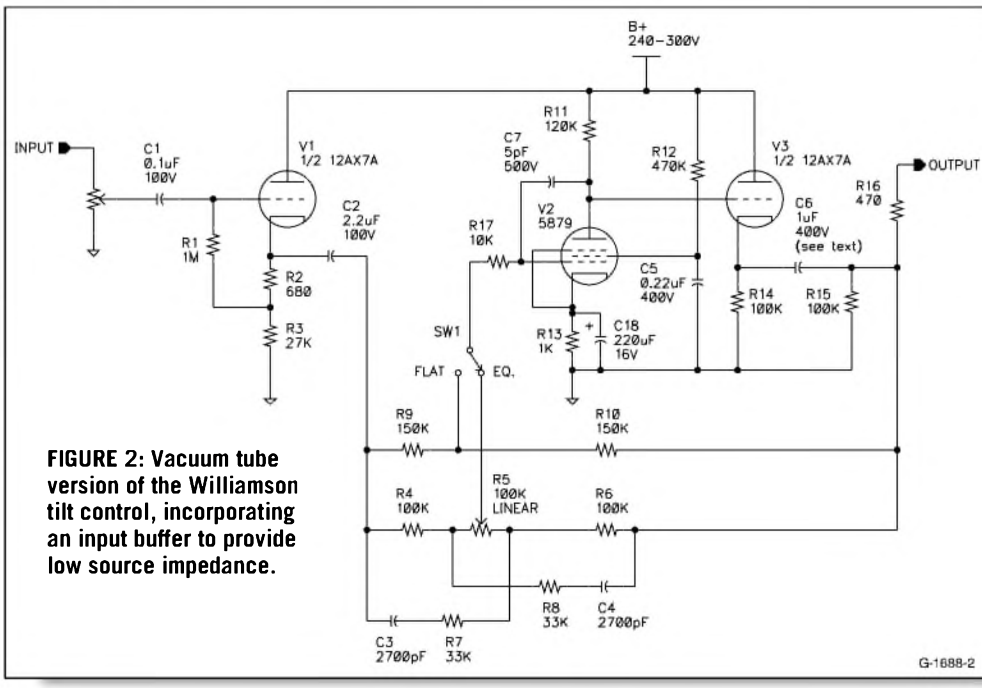


FIGURE 2: Vacuum tube version of the Williamson tilt control, incorporating an input buffer to provide low source impedance.

ABOUT THE AUTHOR

Scott K. Reynolds has a PhD in electrical engineering and is presently employed as an analog and mixed-signal circuit designer at a large computer-systems manufacturer. He became fascinated with electricity in early childhood and learned about vacuum tube electronics from his father's WWII-era textbooks. By the time he was in high school, he was experimenting with his own amplifier circuits. Although his work now involves bipolar and especially CMOS transistors, he'll always like vacuum tubes better. Reynolds lives in the New York City suburbs with his wife and seven-month old son.

Infinite-gain single-amplifier filters get their name from the infinite (that is, very high) open-loop gain of the op-amp, and their design equations are based on the assumption of infinite gain. However, I've found that I obtain results very close to the calculated responses if the open-loop gain is 50 or higher. The tilt circuit will continue to work with still lower open-loop gains, but the response becomes asymmetric about the center frequency, and I observe different amounts of boost and cut.

Since the positive input of the op-amp is grounded, it isn't necessary to have a true op-amp; any high-gain inverting amplifier with high input impedance and low output impedance will work as the active circuit element. Thus, it is easy to implement the circuit with tubes. As Mr. Williamson indicates, you must drive the circuit from a low-impedance source in order for the response to be symmetric.

TUBE VERSION

Figure 2 illustrates a vacuum tube version of the tilt circuit. *Table 1* lists the parts and values. The circuit impedances have been scaled up from those in *Fig. 1* to better match vacuum tube characteristics. A 12AX7 cathode follower (V1) provides a low impedance source to drive the filter circuit. A 5879 pentode (V2) provides the relatively high gain, and a second 12AX7 cathode follower (V3) obtains low output impedance.

The 5879 is a low-noise, medium-gain pentode originally used for microphone preamps. With the circuit values shown, it provides an open-loop gain of about 95 with low distortion. With the feedback circuit in *Fig. 2* disconnected, I applied a 21mV RMS, 1kHz test signal to the 5879 grid and measured second-harmonic distortion of 0.19% at 2V RMS output. No higher-order harmonics were visible above the 0.08% noise floor on my spectrum analyzer.

The circuit in *Fig. 2* will also work without the cathode bypass capacitor C18, in which case the open-loop gain of the 5879 drops to 45 and the distortion (still second harmonic) rises slightly to 0.22% at the same 2V RMS output. High-quality NOS (new, old-stock) 5879s are readily available[4], but other tubes (either medium-gain pentodes or high-gain triodes) could also be used for V2 (with different circuit values).

The circuit in *Fig. 2* forms a complete tilt control, but, as a line amplifier, it has two disadvantages: it inverts the signal, and it provides no closed-loop gain. Both of these shortcomings are remedied by the circuit in *Fig. 3*, which provides a non-inverting gain of 3.2 (10dB). *Table 2* shows the parts. It also adds another useful function, a second-order hiss filter with three selectable cutoff frequencies of 5kHz, 7kHz, and 10kHz.

MEASUREMENTS AND VALUES

In *Fig. 4*, I've plotted the measured frequency response of my prototype of *Fig. 3* for four different settings of the tilt potentiometer (R5), as well as with SW1 in the flat position. All curves in *Fig. 4* were taken with SW2 in the flat position.

The seesaw action of the control is readily apparent. Because of the limited gain of my vacuum tube "op-amp," the curves of *Fig. 4* are not quite as symmetrical as they would be if the circuit were implemented with an IC op-amp, but the degree of symmetry I obtained is more than adequate. *Figure 5* shows the measured frequency response of the hiss filter for each of its cutoff frequencies.

The circuits in *Figs. 2* and *3* are both suitable for driving loads down to about 10kΩ, but the output coupling capacitor C6 should be adjusted for best low-frequency performance. The 1µF value shown is suitable for loads down to about 50kΩ. For 22kΩ loads I recommend 2.2µF, and for 10kΩ loads I recommend 5.1µF.

Solen makes low-cost polypropylene capacitors which are ideal for all coupling capacitors in these circuits. The frequency-determining capacitors in *Figs. 2* and *3* (C3–4 and C8–13) can be dipped mica or polystyrene capacitors with a 50V or higher rating. C7 and C19 can also be dipped mica, with a 500V rating. All resistors can be ½W unless otherwise noted.

Closed-loop distortion in the circuits of *Figs. 2* and *3* is very low. At 2V RMS into

TABLE 1
PARTS LIST FOR *FIG. 2* (EACH CHANNEL)

R1	1MΩ, ½W, 5%
R2	680Ω, ½W, 5%
R3	27kΩ, ½W, 5%
R4, R6, R14–15	100kΩ, ½W, 5%
R5	100kΩ linear taper potentiometer, ½W or more, ganged with the corresponding potentiometer in the opposite channel. (Two single potentiometers are shown in *Photo 1*, which is less convenient.)
R7, R8	33kΩ, ½W, 5%
R9, R10	150kΩ, ½W, 5%
R11	120kΩ, ½W, 5%
R12	470kΩ, ½W, 5%
R13	1kΩ, ½W, 5%
R16	470Ω, ½W, 5%
R17	10kΩ, ½W, 5%
C1	0.1µF, 100V, film
C2	2.2µF, 100V, film
C3–4	2700pF, 50V, dipped mica or polystyrene
C5	0.22µF, 400V, film
C6	1.0µF or more, 400V, film (see text)
C7	5pF, 500V, dipped mica
C18	220µF, 16V, electrolytic
SW1	DPDT switch (To avoid an audible "pop" when throwing this switch, use a shorting or make-before-break rotary switch. Otherwise, a toggle switch as shown in *Photo 1* is adequate.)
V1, V3	½ 12AX7A dual triode (use half for the right channel and half for the left)
V2	5879 pentode

TABLE 2
ADDITIONAL PARTS FOR *FIG. 3* (EACH CHANNEL)

R18–19, 25	100kΩ, ½W, 5%
R20	33kΩ, ½W, 5%
R21–22	330kΩ, ½W, 5%
R23	470kΩ, ½W, 5%
R24	1kΩ, ½W, 5%
R26	10kΩ, ½W, 5%
R27	4.7kΩ, ½W, 5%
R28	120kΩ, ½W, 5%
C8	1000pF, 50V, dipped mica or polystyrene
C9	680pF, 50V, dipped mica or polystyrene
C10	470pF, 50V, dipped mica or polystyrene
C11	100pF, 50V, dipped mica or polystyrene
C12	68pF, 50V, dipped mica or polystyrene
C13	47pF, 50V, dipped mica or polystyrene
C14	0.22µF, 400V, film
C15	1.0µF, 400V, film
C16	33µF, 350V, electrolytic
C17	220µF, 16V, electrolytic
C19	5pF, 500V, dipped mica
SW2	DPDT switch. (To avoid an audible "pop" when throwing this switch, use a shorting or make-before-break rotary switch.)
SW3	4-pole, 3-position shorting (make-before-break) rotary switch, 2 poles used for each channel
V4	5879 pentode
V5	½ 12AX7A dual triode (use half for the right channel and half for the left)

10kΩ, total harmonic distortion is below the 0.08% noise floor of my spectrum analyzer, and intermodulation distortion is below the 0.06% noise floor of my IM analyzer. If you wish to drive resistive loads below 10kΩ or capacitive loads above 1800pF (i.e., more than 60′ of cable at 30pF/foot), or if you need output swings more than a few volts RMS into low impedances like 10kΩ, you may prefer to try higher bias current in the output stage.

Output stage bias current in *Figs. 2* and *3* is about 1.2mA. If you substitute a 12AU7 for V3 and a 22kΩ 3W resistor for R14, output stage bias current will be about 6mA, providing five times the drive current. But there's no need to do this unless you really require high drive current. In fact, I've found that the 12AX7 is a more linear cathode follower, as long as the current drive it provides is sufficient.

Before you commit to the complete line amplifier in *Fig. 3*, I recommend building a prototype of the circuit in *Fig. 2* to see whether you like the action and sound of the tilt control. As Mr. Williamson says, the tilt control is very convenient for adding a bit of warmth to a recording that a recording engineer has made overly bright. You might call it a "warmth control."

CONSTRUCTION

I built my prototypes on perfboard, using PC mount sockets for the tubes (*Photos 1–3*). I inserted the components and tube sockets from one side of the perfboard, just as in printed circuit board construction, and wired the circuit on the back side of the board, using the component leads and short pieces of insulated wire. I often prototype simple circuits this way because it is more compact and less expensive than doing point-to-point wiring in a conventional chassis. I show a component placement diagram in *Fig. 6*, for those readers who wish to make a near-identical copy of my prototype of *Fig. 2*.

For bench testing and listening evaluations, I mounted the perfboard (along with the controls and RCA jacks) on a grounded metal panel salvaged from another project. Either this sort of metal ground plane or a fully enclosed grounded metal box is highly recommended to prevent noise pickup. For safety reasons, some type of enclosure is required to prevent accidental contact with the high voltage nodes in the circuit!

When you build your prototype, don't forget the frequency compensation components R16, C7, and C19, since they are necessary to prevent high-frequency in-

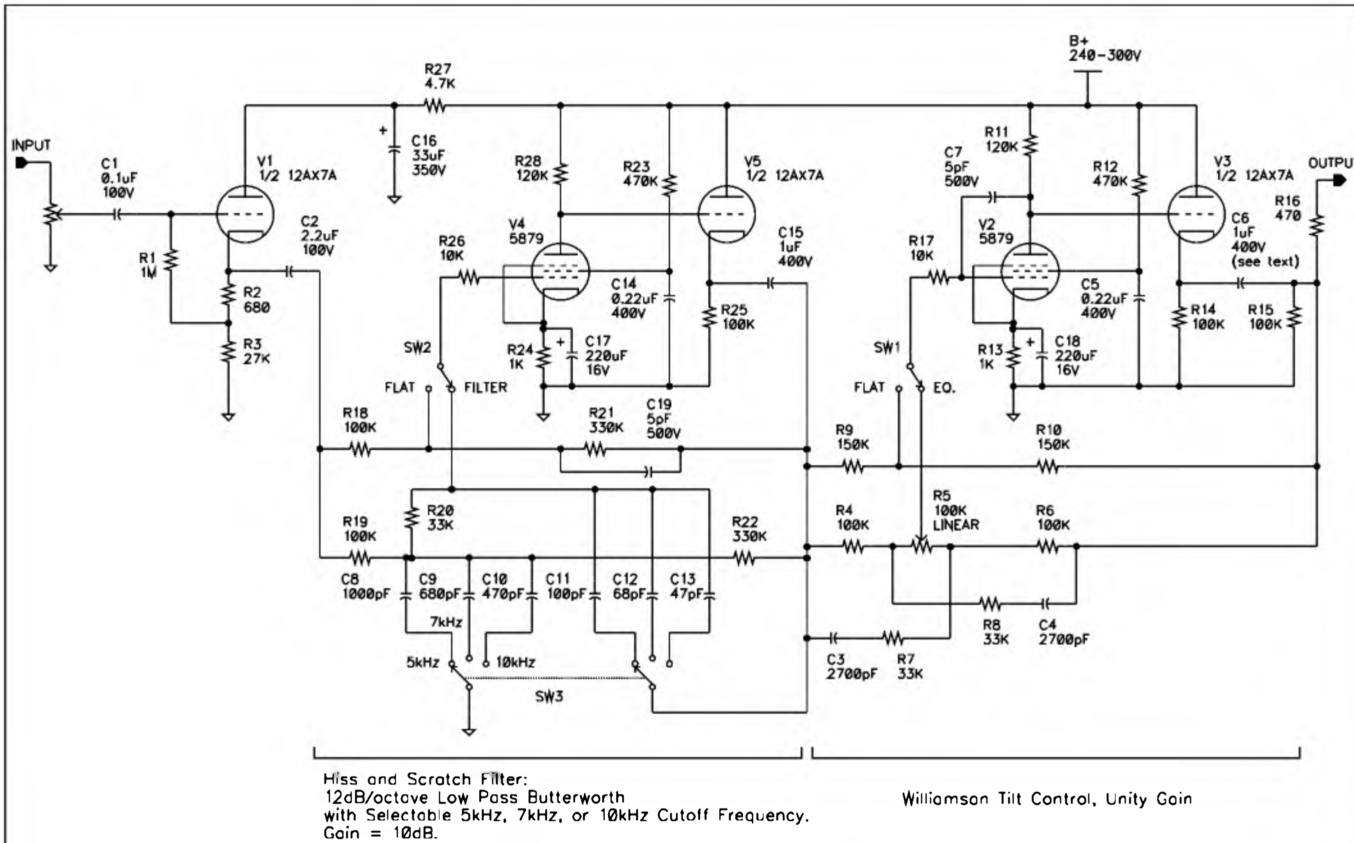


FIGURE 3: Complete line amplifier, incorporating a tilt control and a hiss and scratch filter with selectable cutoff frequencies. This circuit provides a non-inverting gain of 10dB.

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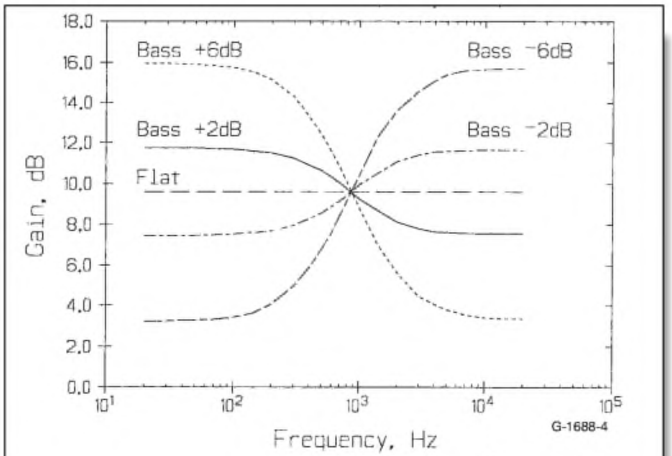


FIGURE 4: Measured frequency response of the tilt control in *Fig. 3* for various settings of the potentiometer R5.

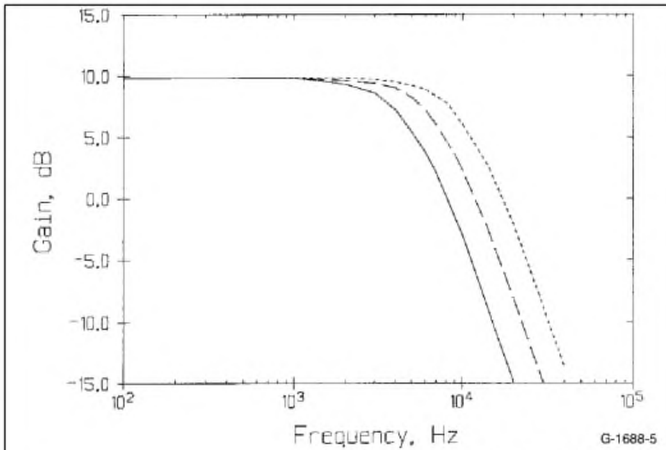


FIGURE 5: Measured frequency response of the hiss filter in *Fig. 3* for each of its three cutoff frequencies (5kHz, 7kHz, and 10kHz).

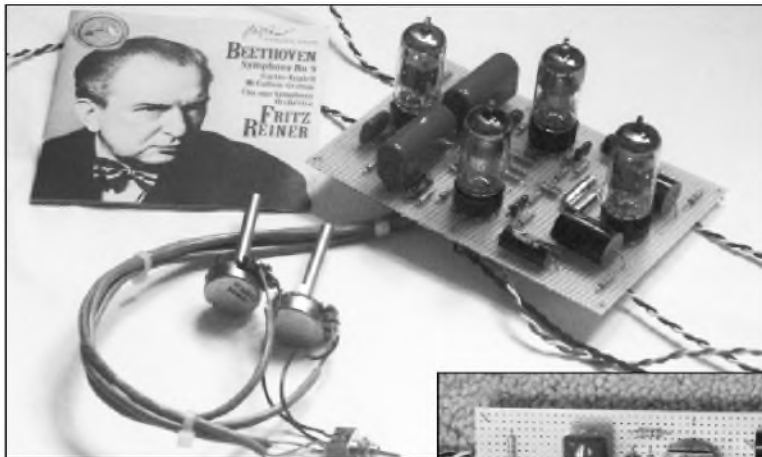
stability under some conditions. The purpose of grid stoppers R17 and R26 is to reduce the chance of radio frequency interference; these should be mounted immediately adjacent to the socket. Finally, if the wires leading to and from the controls SW1-3 and R5 are more than a couple of inches long, you may wish to use shielded cable with the shields connected to ground at one end only (*Photos 1-3*).

If you do use shielded cable, the cables should be no more than 8–10″ long. To keep the wires short, mount all the controls (SW1-3 and R5) directly on the perfboard, with shaft extensions (if necessary) to reach the front panel. (I didn't do this because I didn't think of it until afterward.)

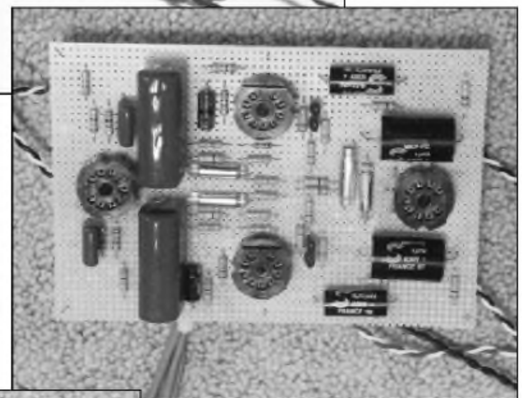
Proper wiring of the ground connections is important in any audio circuit. I made all the circuit grounds to a piece of 14-gauge copper bus wire on the perfboard (*Photo 3* and *Fig. 6*). The input and output jacks should be isolated from the chassis with a nylon washer, and both the ground and signal connections for each jack should be run by way of a shielded cable (or twisted pair, if you prefer) to the perfboard. The metal chassis or panel should be connected at one point to the bus wire on the perfboard. It is easiest to obtain low noise and hum if your power supply is in a separate box.

POWER SUPPLY

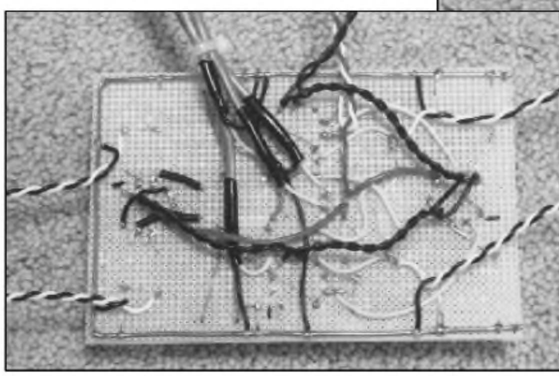
A suitable power supply for either circuit



▲ **PHOTO 1: Prototype of the circuit in *Fig. 2*.**



▶ **PHOTO 2: Top view without tubes showing component placement. Note that C7 and R16 have changed position slightly since this photograph was taken, so the component placement diagram (*Fig. 6*) should be followed for those parts.**



◀ **PHOTO 3: Bottom view showing how the circuit is wired using component leads and short pieces of insulated wire.**

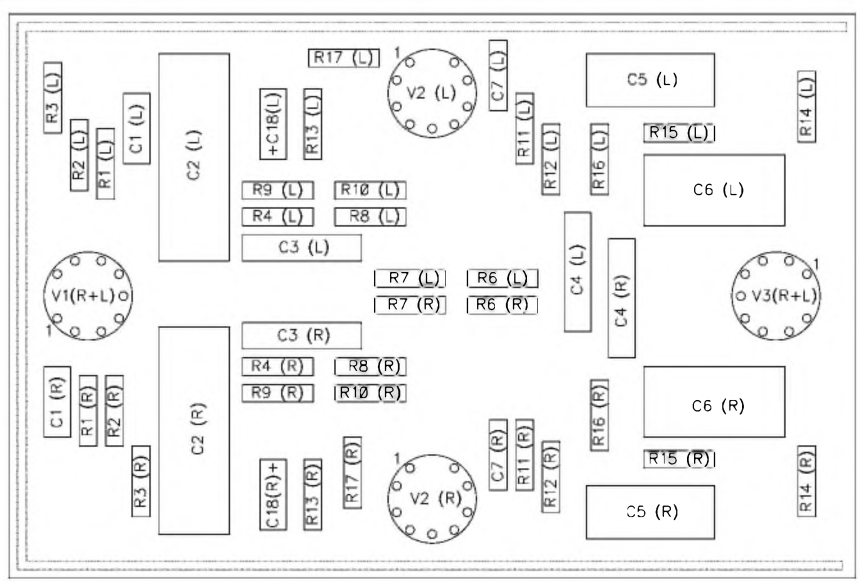


FIGURE 6: Parts placement diagram for the author's prototype of the circuit in *Fig. 2*. This is a top (component side) view corresponding to *Photo 2*. The double lines around three sides of the board show the location of the copper ground wire underneath the board (see *Photo 3*).

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TABLE 3
POWER SUPPLY PARTS LIST (FOR TWO CHANNELS)

R101	3kΩ, 10W, 5%, wirewound
R102	270kΩ, 2W, 5%
R103	39kΩ, ½W, 5%
C101	47µF, 450V, electrolytic
C102	100µF, 450V, electrolytic
C103	50µF, 50V, electrolytic
C104	0.1µF, 400V, film
D101-2	1N4007, 1000V, 1A rectifier diode
V101	0B2, 105V gas-filled regulator
V102	0A2, 150V gas-filled regulator
L101	15H at 75mA filter choke, 411Ω DC resistance, Antique Electronic Supply P-T158L (see text)
T101	250-0-250V at 40mA, 6.3VCT at 2A, Antique Electronic Supply P-T995 (see text)

REFERENCES

1. Aaron Freed, "Letters," *GA* 6/99, p. 70.

2. Reg Williamson and Alan Watling, "A New Control Preamp: Back to the Future," *TAA* 4/91, p. 10.

3. Lawrence P. Huelsman, *Active and Passive Analog Filter Design*, McGraw-Hill (New York), 1993, ISBN #0-07-030860-8, p. 266.

4. I bought some recently from Triode Electronics, (773) 871-7459, www.triodeel.com. They are also available from Antique Electronic Supply.

5. *RCA Receiving Tube Manual*, RC-30, 1975, pp. 112–113.

6. Scott K. Reynolds, "70W 6L6GC Low-Distortion Amp," *GA* 5/99, p. 1.

is shown in *Fig. 7*. *Table 3* is the parts list. It uses a 250-0-250V at 40mA, 6.3V CT at 2A power transformer, which I purchased as surplus from Antique Electronic Supply. I've occasionally seen such small transformers available as surplus from various dealers, so call around before you buy a new one. The filter choke L101 can be anything that will handle the 30mA load current in the range of 5–20H.

The capacitor across the voltage regulator tubes (C104) must be limited to 0.1µF to prevent possible oscillation.[5] If your power supply is far from your audio circuitry, you can place C104 on the same board as the audio circuits instead of in the supply. This more effectively bypasses the power supply at the audio circuitry, where it is important. I recommend a DC heater supply for lowest hum, but this is not absolutely necessary if you just choose to build a quick prototype and "have a listen," as I did. For the more ambitious, a suitable 6.3V DC heater supply was described in one of my earlier articles.[6]

I hope that some *audioXpress* readers will build this tilt control. If you do, let me know how your project turns out and what you think of its sound. I can be reached by e-mail at skreyn1@attglobal.net. ❖

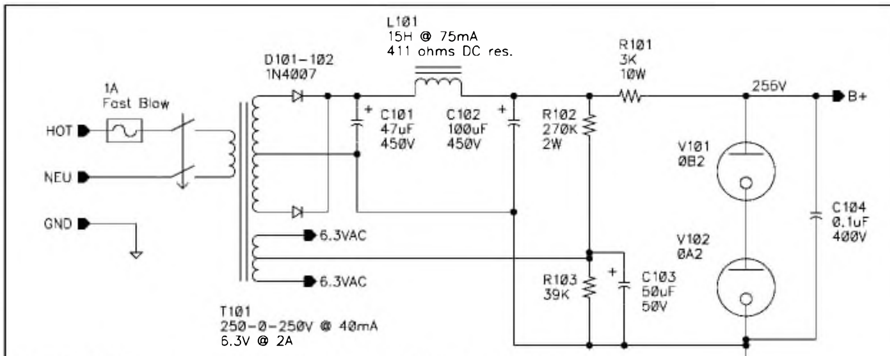

FIGURE 7: Power supply for the circuits of *Figs. 2* or *3*.

A Simple Golden Powered Sub

Here's the second in a series of quick-and-easy projects to hone your speaker-building skills, which you can someday apply to your dream undertaking. **By Lester Mertz**

I had just purchased (for about \$60 from an MCM sale catalog) a 70W subwoofer amplifier with which to experiment. I had planned to use it with a dual-voice-coil (DVC) 8″ unit, but after reading the fine print that came in the box—"4Ω minimum load; no warranty if you blow it up," or words to that effect—I knew it wasn't going to work.

The 8″ DVC woofer was 4Ω on each coil, and the two in parallel meant 2Ω. I tried the coils in series, but the amp did not seem to drive the speaker to realistic levels. It just didn't have the needed power. Then, as I was about to send the amp back, I remembered the two \$10 6″ Martin W-678 woofers I had on hand. They had been in another pair of cabinets, a two-way design, and I had replaced them with upgrades to newer polyfiber units. So now the orphans were just collecting dust, and they were 8Ω each! I decided to give them another life.

PHOTO 1: The box glued and thoroughly clamped.

A SECOND LIFE

Most 6″ drivers work well in a 0.5ft^3 box. I intended to put both speakers and the amp in the same box, which meant 2 × 0.5ft^3, for a total box size of 1ft^3. I didn't worry about the volume lost to the amp or the speakers, and since the box was small, it would need no braces.

Speaker designers like to use terms like "golden mean," "golden section" and other esoteric names, but they're just box-dimension ratios, such as 0.6:1:1.6, or 0.79:1:1.26. The objective of using such ratios is to build an enclosure with dimensions that do not match up with each other in any way that would cause a reinforcement, which simply means an adding of the sounds inside the box. Achieving nonresonance is the aim.

For example, a box 12″ × 24″ × 48″ would not be good, since the dimensions, being exact multiples of one another, would produce that reinforcement of particular frequencies just mentioned as being undesirable. Following is a series of rules of thumb for determining desirable dimensions.

First, 1ft^3 = 12″ × 12″ × 12″ = 1,728in^3. You can use this for a box of any volume—2ft^3, 3.5ft^3, or whatever. In your calculator, enter the volume in cubic feet, multiply by 1728, and you get the volume in cubic inches.

If, for example, you have a 3.5ft^3 box, multiply 3.5 by 1,728, obtaining 6,048in^3. Now find the cube root of 6,048, which is a bit larger than 18″. That's the middle inner dimension of the box. Then, using the ratios 0.6:1:1.6, 18″ × 1.6 = 30″ (the long dimension), and 18″ × 0.6 = 11″ (the short dimension). So your 3.5ft^3 box would have inner dimensions of 11″ × 18″ × 30″.

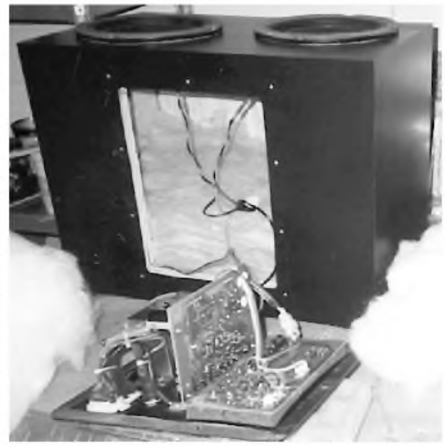

PHOTO 2: Twin subwoofer, with amplifier module in foreground.

Now, using the alternative second ratio, 0.79:1:1.26, the inside dimensions of a 1ft^3 box are approximately 9.5″ × 12″ × 15″. To get the outside dimensions, add the thickness of the material you intend to use, which results in 11″ × 13.5″ × 16.5″ with ¾″ material. [*For another discussion on this subject, see Louis McClure's article, "Determining Optimum Box Dimensions,"* SB *£/00, p. 42. —Ed.*]

BUILDING THE BOX

Looking around, I found some old plywood, and quickly measured it. Not quite enough, but with the amount on hand, the boxes became 10″ × 12″ × 16″ on the outside. This is only about 0.75ft^3 on the inside, but being a tad smaller may not be a bad thing, since it might help with the power handling. Commercial speakers are often mounted in smaller than optimum boxes. Besides the economics (less wood, weight, and shipping costs), this has the advantage of increased power handling because of the compression of the air.

I glued the box together using small finishing nails to hold things in place and then bar clamps to squeeze the glue out of all the joints at once (*Photo 1*). I let it set overnight. The next morning, upon removing the clamps, I saw that two ends were slightly oversized through my quick cutting, so, using the router with a trimming bit, I cut them smooth and clean in about five minutes.

Now I had a completely sealed box, and I took it to the bench where the speakers were, and set them on top. It didn't look quite right, so I turned the box around so the long narrow side was on top, and placed the speakers on it again. Yes, the amplifier would look perfect in the center of the largest panel (*Photo 2*).

Take your square, mark your center lines, measure equal amounts, and pencil in the cutouts. The amp needed an 8.75″ by 6.75″ opening in what would become the back. The "remote control"—a volume control on a wire—would be mounted on the front in a 2.5″ × 3.5625″ opening. For the speakers, you need to cut 5″ circles into the top, offsetting them diagonally to make a good fit.

APPLYING THE FINISH

First drill your start holes. I used a ⅛″ brad-point bit, since it doesn't wobble around while you're getting the bit to bite into the wood. Then use the saber saw. Go slowly here—no mistakes at this point!

◀ **PHOTO 3: Front view of subwoofer, showing "remote volume control."**

▶ **PHOTO 4: Completed subwoofer with protective screens over the drives.**

Once all the holes are cut, decide on a finish. Luckily, I had some big scraps of black Formica®, and, laying them on the panels, I found there was enough. I cut the pieces about ½″ oversized, so they wouldn't need super alignment over the sides.

Working with Formica is very easy—just be sure to allow the adhesive to dry. Coat the back of the material first and place it in front of a fan or in any drafty spot. Then coat your clean plywood enclosure, because the wood soaks up the adhesive, and the laminate does not. Once completely dry, looking milky and with no wet spots, it's ready to stick together.

First, place two dowels across the tacky plywood panel and carefully position the laminate on them. Then carefully remove the dowels, one at a time, with one hand, while pressing the laminate into position with the other. Once it's in place, use a 3″ rubber roller with body-weight pressure to ensure complete contact. When this is done, trim off the excess with the router. Use a mill file, held at a 45° angle, to break the edges all around. If you don't do this, you can get some nasty cuts as you run your hand over the edges.

Find the depressions where you made

the cutouts, drill a big guide hole, and rout out the shapes where you'll mount everything. If you don't wish to laminate, try sanding and spray paint. Use a primer first (it saves a lot of paint), allow it to dry, and then sand and add the finish coat. Sand again if you plan to add a second coat.

MOUNTING THE SPEAKERS

I had originally planned to mount the speakers in a push-pull arrangement, where one faces up, and the other faces down into the box. They are then connected so they both move in the same direction at once. Use a 1.5V battery across each speaker in turn to test them if you are not sure of the polarity. The rationale for push-pull is that most speakers do not move in and out on their suspension in a linear way, but mounting them push-pull tends to cancel this nonlinearity.

However, looking at the old driver's backside was not too appealing, so I mounted them both face up after soldering 10″ red and black leads to each. I used foam weather stripping to make an airtight seal. Then I mounted the volume control on the front (*Photo 3*)—it already had foam seal in place. Plug the connectors together at the amp. Use short, black sheetrock screws, hand-seated, so you don't rip out the MDF. Mark out the amplifier holes and predrill, making sure things are lined up in the center of the cutout.

Then choose some sort of damping material to go inside the box. I prefer using dacron polyester on small sealed boxes, but you can also use fiberglass insulation. Put in enough so that all the interior is filled, but do not pack it tightly. If you have a small scale, you can weigh the damping, using 0.5 lb per cubic foot as a rule of thumb. For this $\frac{3}{4}$ft^3 box, try $\frac{1}{8}$ lb or so. Once you listen to the speakers, you can add or subtract filling, note the differences in the sound, and then make it the way you prefer.

Fish around to find the speaker leads, and match them to the red and black of the amp's output. Strip the wire and use standard wire nuts to connect everything. Then mount the amp and carefully screw it down to ensure a good seal.

FINAL CONNECTIONS

Now you're ready to connect to your system. You can use either RCA interconnects from your line amplifier, or parallel leads from the speaker outputs of the main stereo amp to the terminals on the subamp. I tried both, but the second method seemed to give more signal (volume) to the subwoofer's amp, so that's what I used.

After turning it off, I simply placed the new dual subwoofer on top of my approximately 4ft^3 box (see "Bessel Box Subwoofer," *SB* 2/99) and made a judgment call on the level settings. I put on a CD and sat back. After a few adjustments, everything was blending right in. The used speakers did not require any break-in period, but sounded good right off the bat (*Photo 4*).

The sound does not go as deep as the bigger unit, but it does fill in smoothly after a few adjustments of the volume control. There's a good, solid response in the 60–160Hz range, and this is where most smaller stand-mounted speakers begin to roll off in the bass, because of their distance from the floor, and usually from the side walls, too.

The little dual woofers worked their magic. They are very easy to live with, and they don't set the house to vibrating like the bigger subs. This dual-woofer/amp setup might be worth considering if you're an apartment dweller with neighbors who don't appreciate your musical taste. This is not an over-the-top, blow-you-out-of-the-room "THX certified" subwoofer; it just plays music. ❖

Audio Classroom

Designing Your Own Amplifier: Part 6a: Special Output Circuits

By Norman H. Crowhurst

This article appeared originally in Audiocraft, *February 1957. ©1957 by Audiocom, Inc.*

The part of amplifier design that has always posed the biggest problem is the output circuit—how to get enough power with low distortion and a satisfactory value of source resistance, or damping factor. To illustrate this problem in more detail, we will try various methods of connection for the 5881 tube as a basis for comparison.

TRIODE CONNECTION

Consider first the straight triode connection shown in *Fig. 1*, with a plate voltage of 400, a fixed bias of 45V, and a plate-to-plate load of 4,000Ω. This is the condition known as Class AB1, or "low-loading." The reason for the second name is that the load resistance is much lower than the original value intended for Class A operation; its advantage is that a greater maximum output can be obtained.

Figure 2 shows the load line applied to the characteristic curves of a single tube. The solid straight line represents a resistance of 1,000Ω, which is how 4,000Ω plate-to-plate appears to one tube; the dashed curved line represents the load line for this tube, the other tube being complementary so as to produce the straight line as a composite. From the composite line, we can obtain plate-voltage swings (for positive grid excursions from the 45V bias point) of 48V, 104V, and 162V, respectively, for grid swings of 15V, 30V, and 45V.

A peak plate swing of 162V across the load resistance of 1,000Ω represents an average power output of

$$\frac{(162)^2}{2\times 1{,}000}=13.1\text{W}$$

The tube manual quotes a value of 13.3W for the same operating conditions, which is near enough to the same answer.

A mathematical analysis of this amplification, based on the three points obtained, gives the distortion figures as 3.3% third and 1.25% fifth harmonic. The method of deriving these figures involves trigonometry and is too involved to go into in detail here. Taking the RMS combination of these two components, the total harmonic distortion will be $\sqrt{(3.3)^2 + (1.25)^2}$ = about 3.5%. The tube manual quotes 4.4% for 13.3W. Later in this article we will show a simple graphic method of estimating the amount of distortion.

We can calculate the source resistance from the slope of the –45V curve where it intersects the dashed load line. If we draw a tangent through this point of the curve it will pass through 340V at 0mA, and 620V at 100mA. This represents a resistance of 2.8k. As the slope of the composite curve will be twice as steep at this point, this will become 1.4k at the center point on the load line. This is referred to one half of the output winding and, referred to the 4,000Ω plate-to-plate, would be four times this value, or 5.6k. So the source resistance, in this case, is about 1.4 times the load resistance.

To produce an output stage that is acceptable by modern standards, we must reduce the amount of distortion (quoted as 4.4%), and also the source resistance. This output stage requires a pk-pk swing on each grid of 90V, for which purpose the previous stage should also be of a push-pull variety, using a low-mu twin triode. Therefore, the grid drive to this previous stage, without feedback, must be of pk-pk amplitude somewhere between 5 and 10V.

To obtain adequate feedback over the two stages (which is recognized to be an ideal arrangement from a stability standpoint), we must inject some 10 to 50V pk-pk in the cathode circuit of this drive tube, according to the amount of feedback considered desirable. The cathode resistor of this stage, across which the feedback voltage must be produced, is probably on the order of 2k.

If we take the feedback from the plate circuit of the output tubes, the resistors necessary to produce such a large voltage on the cathode of the drive stage must be physically large enough to dissipate 1 or 2W. More important, the output tubes will no longer have 13W available for the external circuit, because 1 or 2W are expended in the feedback.

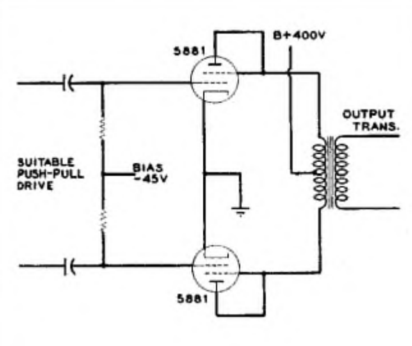

FIGURE 1: Schematic of output circuit, using two 5881 tubes as triodes. Note that screens are connected to plates.

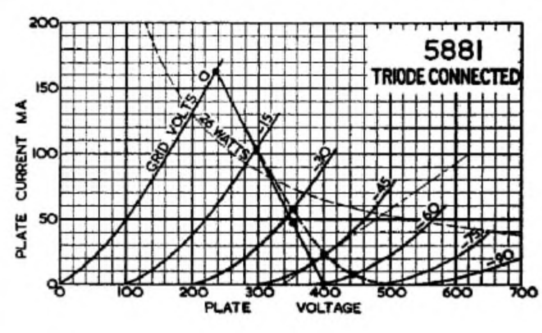

FIGURE 2: Characteristics of 5881 tube triode-connected, with load line to illustrate this method of operation. The solid straight line is one half of the composite line for a plate-to-plate resistance of 4,000Ω. The dotted curved line represents the load component of this composite on the single characteristics shown. Dots on the composite line are projected downward from the intersection with the curved line. The curved line is plotted in the manner described in Part 4a (*GA* 3/00). Also shown is the maximum dissipation curve at 26W.

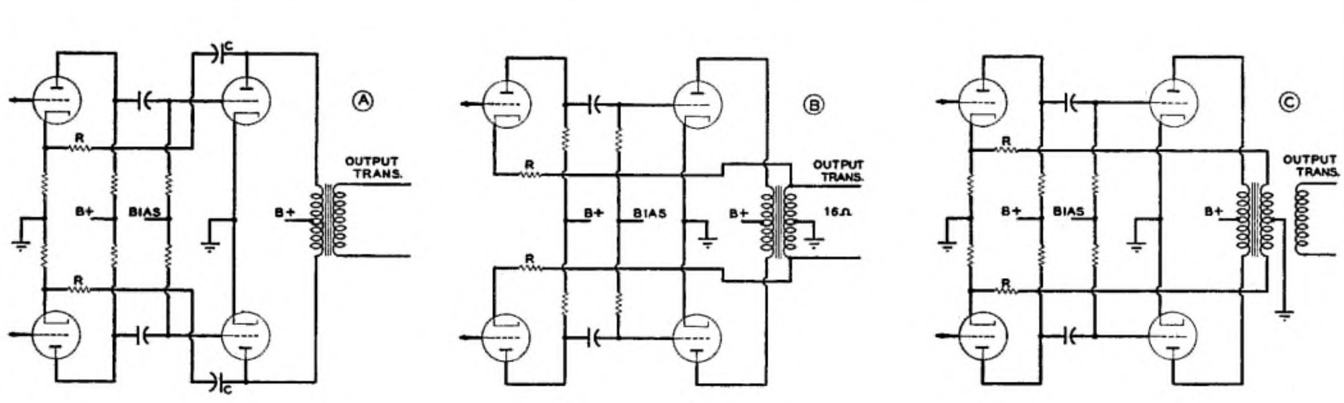

FIGURE 3: Different methods of applying feedback over two stages: at A, the feedback is taken from the plates of the output tubes, requiring a relatively large value of the blocking capacitor, C. Resistor R must dissipate a considerable proportion of the output power to achieve the desired amount of feedback. At B, the feedback is taken from the 16Ω winding of the output transformer, which is here center-tapped. Resistors R may be necessary to provide the correct bias for the drive stage, but will not contribute to the feedback performance appreciably. C is a better arrangement than either A or B, but this requires a special output transformer.

If we consider taking the feedback from the transformer secondary, using the 16Ω winding, which is usually the highest impedance available, the pk-pk voltage developed by 13W across 16Ω is just over 40V. This may be sufficient for the purpose, but it is evident that no series resistor can be used, so the 16Ω winding must be directly coupled to the cathode of the drive stage. This, too, can involve undesirable complications.

The problem appears to arise with triodes because the plate-circuit impedance is too high, and the secondary-circuit impedance is too low. A compromise some manufacturers have adopted utilizes a special feedback winding with an intermediate number of turns. This obviates the necessity for the large-value blocking capacitor, and reduces the power dissipation involved in feeding back from the plate circuit, yet permits use of some isolating resistance.

The foregoing comparison is illustrated in *Fig. 3*. An obvious snag is that a special transformer must be procured. Consequently, this method is readily available only to manufacturers having facilities for producing transformers with custom specifications. For that reason, we will not pursue the discussion of this arrangement further.

Before leaving it, however, we may add that the method is equally applicable to pentode or other methods of connection as output stages. As might be expected, because the pentode has a higher gain than a triode, the previous stage needs to give much less swing to obtain maximum output, and this in turn means that a smaller voltage, tapped off from the output, will give a corresponding amount of feedback. What offsets this fact is the high plate resistance of the pentode, which means that a much larger amount of feedback must be used to obtain a satisfactory damping factor.

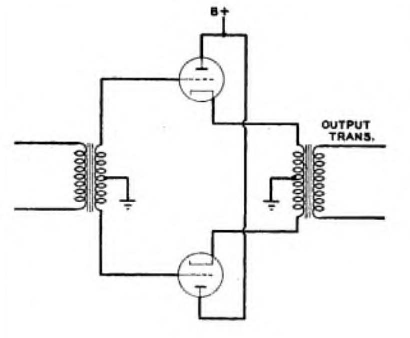

FIGURE 4: Use of triode tubes in push-pull cathode-follower connection. As shown, the tubes are assumed to be biased by the DC resistance of the output-transformer primary halves. This may need additional resistance, or fixed bias can be applied to the center tap of the drive transformer, instead of returning it to ground. A drive transformer is the only simple way of getting sufficient swing for a cathode-follower stage without using a higher B+ for the drive stage.

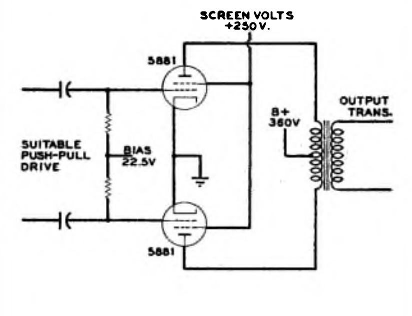

FIGURE 5: Schematic of output circuit, using 5881 tubes as tetrodes (commonly called pentodes, because of the similarity of the operating characteristics).

CATHODE-FOLLOWER OUTPUT

The next circuit we shall consider here is the cathode follower, because this has several variants that we can discuss later in the article. *Figure 4* shows the arrangement of the cathode follower for a triode output.

Since the plate-voltage swing is now developed across a load connected in the cathode circuit, the grid must be driven by a voltage made up of the former plate-voltage swing added to the normal grid swing given by the load line. This is a total swing in each direction of 162 + 45 = 207V: a pk-pk drive of 414V for each tube. A drive stage using our assumed 400V plate supply would obviously need a transformer to produce this much voltage.

The stage now requires a grid-voltage swing of 207V to produce a grid-to-cathode swing of 45V, which represents a feedback ratio of 4.6. Thus the original distortion figure of 4.4% will be reduced to less than 1%.

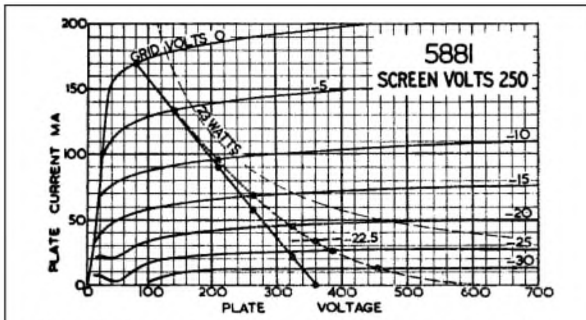

FIGURE 6: Characteristics for the 5881 tube operation in the circuit of Fig. 5. Method of derivation is similar to that in Fig. 2.

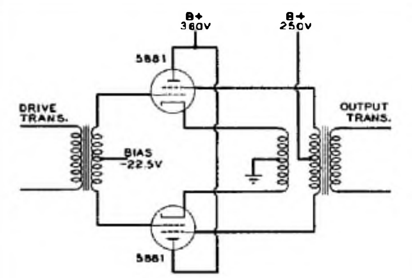

FIGURE 7: This circuit arrangement can be used to operate the output tubes as pentode (strictly, tetrode) cathode followers, with the same operating conditions as shown in the load line of Fig. 6.

Now, how about the source resistance? Consider the operating point for 15V positive swing; that is, the −30V point on the composite load line. This gives 354 plate volts at 48mA. A negative grid swing of 15V from this point brings us to the operating point where the resultant plate current is zero.

Assuming that the actual volts on the grid are maintained the same (that this change of condition is brought about by a change of cathode voltage) the plate-to-cathode voltage will now have dropped 15V, from 354V to 339V. Referred to the 1,000Ω load impedance presented by one half of the output winding, the source resistance gives a change of 15V for 48mA, representing about 312Ω. This is 1,250Ω cathode-to-cathode; the source resistance is a little less than ¼ the load.

PENTODE CONNECTION

Figure 5 shows a pentode connection of this same tube. The composite load condition for one tube is shown in *Fig. 6*, assuming a screen voltage of 250 with a plate voltage of 360. The nearest values to this given in the tube manual listing are a plate voltage of 360 and a screen voltage of 270, with the same bias voltage of –22.5. Tube-characteristic curves for a screen voltage of 270 are not available, so we compute the results from the tube curves for the screen voltage of 250, using the plate-to-plate resistance of 6.6k quoted in the tube manual.

The load-line slope is drawn at 1.65k.

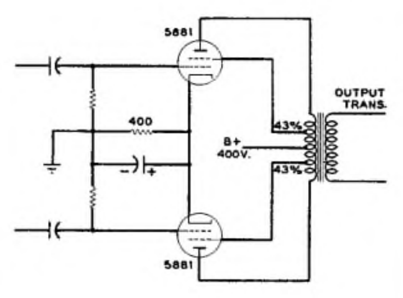

FIGURE 8: Schematic of circuit for operating 5881 tubes in the ultralinear circuit. This particular arrangement uses cathode bias, but fixed bias can also be used. As explained in the text, the ultralinear operation is not so critical.

Peak plate-voltage swing is 280, which represents an average output of 23.7W, compared to the 26.5W quoted in the manual. This seems to be consistent with the difference in screen voltage from 250 to 270.

A mathematical computation from the points on the given load line yields harmonic distortion figures of 2.5% third and 1.5% seventh, a total of 2.9%. The tube manual gives 2% for the screen voltage of 270.

To quote a source resistance for this form of output is somewhat meaningless, because the slope of the tube characteristics varies all the way up the load line. We can, however, apply the cathode-follower connection to the pentode circuit by the arrangement shown in *Fig. 7*, in which case the feedback tends to hold the source resistance more constant. Consider a 2.5V change near the operating point. From the characteristics, it may be seen that this gives a current change of 22mA, for a voltage change of 2.5V, representing an impedance of about 114Ω, or 450Ω cathode-to-cathode.

In this case the input swing required is 280V cathode swing, plus 22.5V grid-to-cathode swing, giving a total of 305, or 610 pk-pk for each tube. This is an even larger requirement than for triode operation. But since we are using a 305V peak to obtain a 22.5V effective swing, the feedback ratio is 13.5. This means that the distortion figure of 2.9% will be reduced to about 0.22%.

To operate a pentode as a cathode follower, the screen voltage must swing exactly with the cathode voltage to keep the cathode-to-screen potential constant. In a voltage-amplifying stage, this could probably be accomplished by decoupling the screen to cathode rather than ground, but in a power stage the only feasible way to achieve this end is by using a transformer with an extra winding, as shown in *Fig. 7.*

The cathode and screen windings should be extremely tightly coupled to avoid any difference in voltage between cathode and screen at any point in the frequency spectrum. A difference in voltage at this point would result in serious distortion, because of the extreme nonlinearity of screen current with screen voltage. To the best of my knowledge, the cathode-follower pentode output is not used. But we will consider later a circuit that is used, and which may be considered to be derived from it.

ULTRALINEAR CIRCUIT

The third operating condition for output tubes is becoming popular for good reason. From discussions of triode and pentode operation, it seems that neither is exactly ideal. It is obvious that a method of operation lying somewhere between these two extremes might be optimal. This is precisely what the ultralinear or tapped-screen method of operation achieves.

The circuit is shown in *Fig. 8.* Static tube characteristics for this connection, as published by Tung-Sol, are illustrated in *Fig. 9.*

Ultralinear operation is fairly simple to understand by comparison with triode and pentode operation. In a pentode circuit, the screen voltage is held constant, while the plate voltage is allowed to swing according to the grid-voltage drive. In a triode-connected pentode, the screen is strapped to the plate; hence, its voltage swings by precisely the same amount as

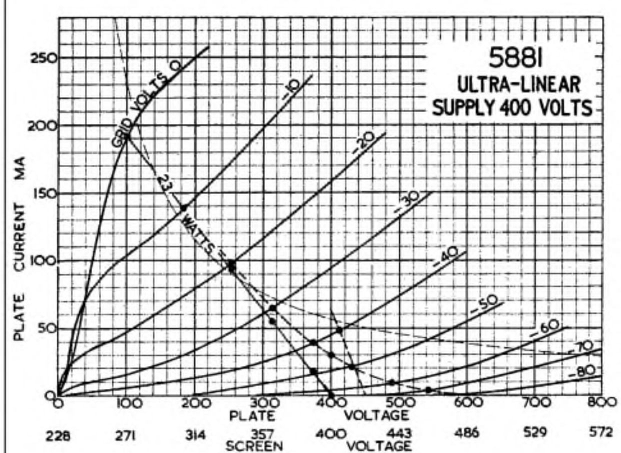

FIGURE 9: Tube characteristics for ultralinear operation with a supply of 400V (plate or screen to cathode). The curves were plotted by reducing the screen voltage from 400 by 43% of the reduction used for the plate (increase same way). Composite and individual load lines are also shown, as for triode and pentode operation.

the plate voltage. Ultralinear operation splits the difference and, in the case of the curves shown in *Fig. 9*, the screen voltage swings by 43% of the plate voltage.

This is achieved in practice by using 43% taps on the plate winding for connecting the screens. The value of the optimal tapping for different tube types will vary somewhat, but 43% has been found ideal for quite a variety of tubes. The ultralinear circuit (*Fig. 8*) uses automatic bias, so the bias will change somewhat from zero to full output.

The sloping dashed line in *Fig. 9* shows the condition for zero signal. The slope of this line represents a resistance of 800Ω. Bias-resistor value is actually 400Ω, but since the current from both tubes passes through it, this can be regarded as representing 800Ω per tube. The quiescent operating position is thus found to be about −40V bias, with almost 50mA plate current in each tube.

Bias at the maximum drive condition rises to about −45V. The plate current swings from 30mA per tube, or a 60mA total (where the cycle passes through the operating point), to a peak of about 190mA, giving an average of about 115mA. This produces 46V across 400Ω.

The solid load line in *Fig. 9* represents a plate-to-plate resistance of 6.5k (or a resistance for the single plate of 1.625k). The peak plate-voltage swing, 300V, produces an output power of 28W, which is higher than either the pentode or triode method of operation.

Distortion is found to be about 3.3%, almost pure third-harmonic. It may be wondered at this point why such a method of operation is called "ultralinear," since the distortion is not appreciably better than either of the other methods of operation.

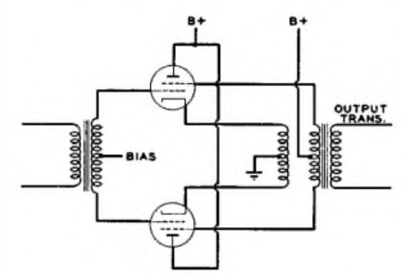

FIGURE 10: Schematic of circuit for operating tubes as ultralinear cathode followers. For 43% equivalent tapping, the screen winding must have 57% of the turns in the cathode winding. Extremely good coupling between the cathode and screen windings is essential.

What the simple figure of distortion does not indicate is how tolerant the circuit is of different impedances.

Compared with triode operation, this arrangement gives more than twice the power. Compared with pentode operation, the increase in power is not so great, but the performance is almost as independent of load as with triode operation. The composite characteristics, if drawn in full, would prove to be much nearer to a set of parallel straight lines than any of the other methods of operation, with the result that this circuit is not critical of inductance components in the load. That is very important in the practical working of the circuit, although it may not be so important when the amplifier is tested with a resistance load.

It would, of course, be possible to operate the ultralinear circuit as a cathode follower, by so coupling the screen that its voltage swings by 57% of the cathode swing while the plate voltage is connected to B+, as shown in *Fig. 10.* This would require an input drive of 300 + 45 = 345V peak, or 690V pk-pk, which is even more than the drive required for pentode operation. The feedback ratio is 345/45 or 7.67, which would reduce the distortion to less than 0.5%

The source resistance as a cathode follower can be obtained by considering 5V excursion from the operating point at full output. This is 18mA for 5V, or about 280Ω. There would be about 1,100Ω cathode-to-cathode, or about one-sixth of the load resistance. The ultralinear circuit operating normally will have over 7½ times this value, or about 1.25 times the load resistance. As described in Part 5 (*GA* 6/00), this would be a good starting point for applying overall feedback, and it would be quite permissible to use, say, 20dB of overall feedback to achieve a damping factor of about 8. ❖

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IMPROVING A CLASSIC

"The Future of Vacuum Tubes in Audio, Part 1" (*GA* 3/00, p. 20) refers to the highly popular Dynaco Stereo 70 power amplifier. Considering its simplicity, it is a fine unit. However, a few simple changes may improve it significantly.

First, the 82pF capacitor in the 7199 pentode plate circuit causes the open loop gain to roll (–3dB) at about 7 to 8kHz. If high-frequency stability does not suffer, reducing this capacitor to 27pF or 33pF, moving the –3dB point above the audible range, may give clearer highs. You may also need to change the 390pF capacitor between the lower output tube screen grid and the input tube cathode circuit for best high-frequency transient response.

Second, it would be a good idea to provide for measuring and balancing the individual output tube cathode currents. This is important in any amplifier that uses fixed bias. Individual 27Ω cathode resistors, with a removable jumper to tie the cathodes back together during normal listening use—plus a balance control network where the 270k grid resistors connect to the bias supply—would do the trick.

One concern I have about the Stereo 70 is the single GZ34/5AR4 rectifier tube, which has a load current rating of 250mA, used to power two pairs of EL34/6CA7s. Each pair draws 100–120mA idling and 170–180mA at full output, and each 7199 draws 4–5mA. Adding this up and doubling for two channels, the 5AR4 is running near full-rated load current at idle and would be overloaded if the amplifier is being run full blast.

If the power transformer high-voltage winding is rated to handle the full output current, silicon rectifiers with a delayed soft start B+ turn-on would be more reliable. Commercial-duty sound amplifiers using four 6CA7s generally had two 5U4GBs in parallel.

One commendable feature of the Dyna Stereo 70/Mark IV is the absence of an electrolytic bypass capacitor across the 620Ω 7199 pentode cathode bias resistor.

Michael Kiley
Crestwood, Ill.

TUBE IDENTITY

I have constructed the dynamic headphone amp described in the premier issue of *Glass Audio* (1988), which has all the clarity that the authors described. A friend of mine auditioned the amp with Sara McLachlan's *Surfacing* CD. His response was, "I have never heard a cello up close before!"

The design calls for a PCL805, which is identical to the ECL805 except the heater voltage is 18. I could find only four PCL805s and I purchased many PCL85s. I could hear no difference between them. According to the RCA tube manual (RC30), the PCL85 does not exceed any ratings in this circuit. The RCA tube manual does not list the PCL805. I have used the PCL85s for a year now without any difficulty, but I would like to know what the differences are between the PCL805 and PCL85.

Jim Dungan
Port Neches, Tex.

Rickard Berglund responds:

The Philips tube manual states that the PCL85 is identical to the PCL805.

PERFORMANCE MEASURE

I ordered a couple of back issues of *GA* to review some amplifier designs, and also added a new subscription. The articles were great except for one critical area. They didn't have much in the way of performance testing. Many of these amplifiers may cost $2500 or more to build a pair (monoblocks), not to mention the time involved. To go through that expense and time of building them without really knowing what the outcome will be seems pretty risky for anyone but the most wealthy and those with the most free time.

May I suggest that your authors include a minimum set of specifications such as: frequency response at 1W and full power; distortion at 1W and full power at, say, 20Hz, 1kHz, and 20kHz; output impedance; and power consumption at idle and full power. I am sure you have more ideas about what could be included.

Considering the time it takes the author to build the amplifiers, the time it would take to measure its performance would be minuscule. I think this would not only be helpful to your readers, but also enhance the quality and credibility of your publication.

Michael Adams
Pacific Palisade, Calif.

SURROUND REPAIR

I own four JBL136A 15″ woofers purchased in the mid-1970s. About 12 years ago the surrounds turned to powder. I had them re-coned at considerable expense. I was told at that time that the material that JBL used was susceptible to age, humidity, smog, ozone layer depletion, and so forth. Sure enough, it has happened again.

Is this repair a procedure that I can do at home, or should I leave it to a professional? Does anyone make a surround that has a longer life?

Clif Penick
Northport, Ala.

Contact Image Communications at Dave_Armon.woodsind.com for companies who sell surround repair kits for a wide variety of drivers.—Ed.

MONARCHY UPGRADE

I just finished reading Gary Galo's article on upgrading the Monarchy DAC (*AE* 2/99) and had a couple of comments. The first is that while this mod may work for the parts specified, this does not mean that you can apply it universally.

The Analog Devices (ADI) D/A converter chip AD1860 was mentioned in the article. There should be a warning. The AD1860 includes an uncommitted op amp designed to be used as an I/V converter. It is not good practice to connect the output of op amps together. If you are going to do this mod with the ADI parts, you should not parallel pins 9, 10, and 11 to the existing IC.

Instead, bend them so they are parallel to the body of the IC. Then jumper the op amp output (pin 9) to the inverting input (pin 11). This completes the feedback loop around the op amp and basically takes it out of the picture. If you don't make this connection, the op amp will be running open circuit and could inject noise into the system.

If you were to just hook up the second IC, you may not see any problems. After all, the outputs of the two op amps should be very close to each other and, therefore, shouldn't really be fighting each other. But for top-end performance, why allow the potential for problems?

The second issue is with the trim: specifically, the trim adjusting out of the differential nonlinearity of the MSB. By paralleling the ICs, you limit how close you can adjust out this error. You should always check the adjustment. Just because it was set for one D/A chip does not mean that it will be correct for two.

The differences, however, should be minimal. You may need to revisit these adjustments from time to time because of changes due to mechanical vibration, heat, and the like changing the pot setting. Again, the differences will be minor, but real.

Note that the PCM63 also includes this adjustment. It just isn't included in most designs. Check Fig. 5 of the PCM63 data sheet.

Hank Zumbahlen
Boulder Creek, Calif.

Gary Galo responds:

Hank's points are well-taken, and should be heeded by those who try this modification. Both Monarchy and I ignored the internal I/V converter, since it is not used in the 18B. When Bob Adams of Analog Devices replied to my inquiry regarding the paralleling of the AD1860 DACs, he did not mention these caveats. In his defense, he may have assumed that I was already aware of them, and perhaps I should have been.

Regarding the trim, I agree that it is best to check the trim if test equipment is available, and I pointed this out in the article. In my sample of the Monarchy Model 18B, there was no change in linearity when the second DAC chips were paralleled. But, this may not hold true for every sample.

Burr-Brown does include a provision for external trimming of the PCM63, but its implementation is relatively complex, requiring two external fixed resistors, two trimpots, and two capacitors. As the PCM63 data sheet notes, "Great care should be taken, however, as improper adjustment will easily result in degraded performance." The performance of the PCM63 is so good without external trimming that most manufacturers who use this chip seem to have concluded that the trim procedure is not worth the added expense. Unless the trimming is implemented and adjusted with the greatest of care, performance will actually be worse.

TIME DELAY

I am considering building a pair of the 100W triode amps featured in *GA* 3/00. Are there any corrections to the schematics as printed? Would the soft-start circuit using the 6X4 ("Glass Shards," *GA* 3/00, p. 70) be appropriate for the amp?

Charles Gutzman
Petersburg, Ill.

Joseph Norwood Still responds:

Good luck on your 100W project, which is a big one but should prove to be a lot of fun and will certainly give hours of great listening pleasure.

Per changes: Capacitors C5 and C6 shown on the parts list as 0.33µF are correct. The schematic diagram (0.15µF) is incorrect. The 0.33µF value improves the 100Hz square wave. (Be sure to use Orange Caps.) Capacitors C6 and C7 on the power-supply schematic should be 47nF, while .047µF on the parts list is correct. "Available from Antique Electronic Supply" should appear between S1 and R3 on the parts list.

The soft-start circuit in the same issue is a novel circuit but would not be able to "handle" the current of the 6-6550s. I recommend a conventional relay with a time-delay circuit. A time-delay device manufactured by CEBEK is available from MCM Electronics (1-800-543-4335), fully assembled, for $19.95, FN28-5100.

The time delay is adjustable from 1-180 seconds and has relay contacts rated at 5A. An external 12V DC, 50mA power supply is required to power and activate the timer, which is deactivated when power is removed. You may purchase the power supply separately or build the one in my article for about $9; all parts are from Radio Shack. The only modification to my power supply is that you must use a 12V regulator (7812) instead of the 9V regulator (7809).

I put my "head" and soldering iron to work and came up with a simple, inexpensive time-delay relay (Fig. 1). Table 1 *is the parts list. You must be careful when using MOSFETs; first mount all the components on the PC board, then mount the IFR510. Solder the source, then drain and finally gate to the appropriate location. Make sure all capacitors are discharged prior to mounting the IFR510. Note: When you switch on the IFR510 (after 20 seconds), expect +3V DC at the junction of C3 and R2 and also 0.1V DC at the drain of IFR510.*

The cost differential between my unit and the purchased unit is the cost of the 12V DC 50mA supply required to "energize" the commercial unit. The CEBEK unit is certainly more sophisticated than my simple design. I leave it up to you as to which unit you prefer. Personally, I would go with the commercial unit.

TABLE 1
MATERIALS

QUANTITY	REFERENCE	PART
1	B1	100Vpiv/1.4A
1	C1	1000µF
1	C2	100N
1	C3	100µF
1	F1	Fuse
1	IC1	LM7809
1	Q1	IFR510
1	RE1	9V/500Ω
	12A contacts, RS 275-005A	
1	R1	220k
1	R2	120k
1	TR1	12.6V, 300mA
1	PC board RS 276-158B	

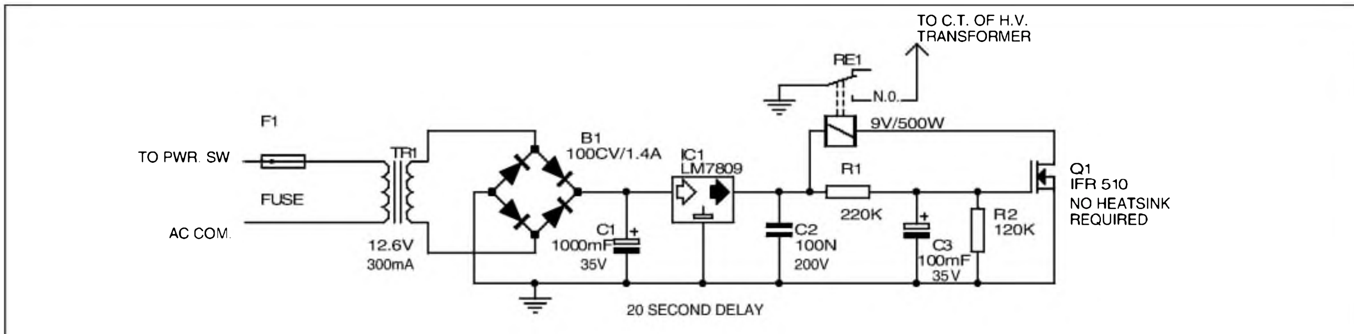


FIGURE 1: A simple time-delay relay, total cost is $18.50, and all parts are available from Radio Shack. G-Still-1

I can understand your concern with "cathode stripping," since many other readers of GA *have requested information on a time-delay circuit. I hope this clarifies the issue and makes everyone aware that the "bare-bones" cost of a time-delay device is $20.*

Again, good luck on your project. I suggest this fall or the winter would be an appropriate time for such a massive project. The amplifier has a great sound, is very stable, and runs quite "cool" (with the aid of the 65 CFM blower fan). The excellent 10kHz square wave of the amplifier is attributed to the low impedance, high-current operation of the 6550s and the excellent design of the Hammond transformers—and, of course, the triode configurations.

TUBE & GROUND DATA

In *GA* 2/00, you printed a letter from John Badalamenti inquiring about amplifiers using 807 output tubes (p. 67). Eric Barbour's reply indicated that a pair of 807s could not reliably be driven to 100W. I am submitting a schematic for a Bogen HO-125 amplifier, made around 1950, which used a pair of 807s to obtain 125W (*Fig. 2*). This unit ran the 807s at 840V, and used parallel 6SN7s as drivers. I do not know the distortion ratings. While this was a commercial amplifier, there was not much distinction between hi-fi and commercial amplifiers in 1950.

From my experience, 807s are very rugged tubes. While it is true that they were rated at 600 plate volts, this was very conservative. Originally, 807s were used as transmitting tubes. Later, they appeared in receiving tube manuals with specifications for use as Class AB-1 amplifiers. Apparently, they can be used at higher plate voltages and dissipations for audio applications. New and used 807s can easily be found at hamfests.

Also, Mike Gustafson's power supply schematic in the same issue (p. 32) shows what I believe to be a very dangerous error. The indicator lamp is connected from the hot side of the AC line to ground. The lamp will work, but only if the unit is plugged into a grounded three-wire outlet. If plugged into an ungrounded or two-wire outlet, 110V AC will appear on the chassis. The ground wire should never be used as a current-carrying conductor. Its purpose is

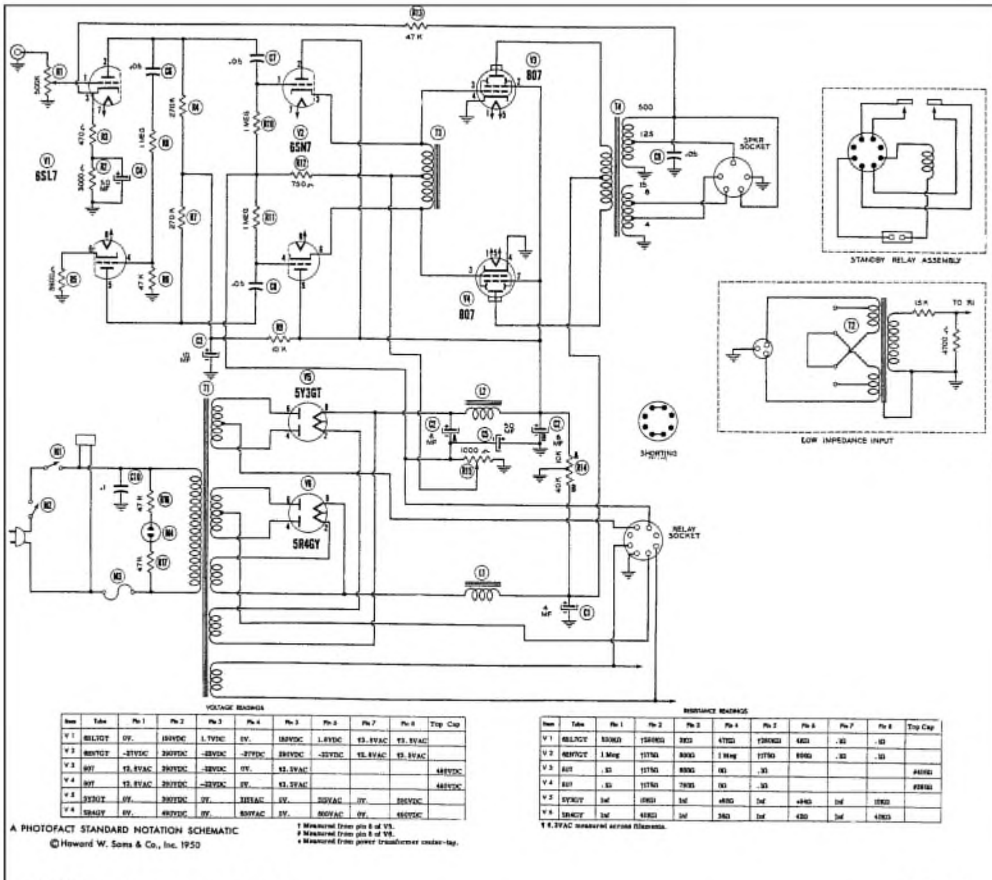


FIGURE 2: Bogen HO-125 amp.

to carry only fault current, as in the case of an accidental hot wire to chassis short.

If wired as shown, you could receive a nasty shock (through the indicator lamp) if the unit is not grounded and you touch it while you are grounded. If the lamp has high enough wattage, the shock could be fatal. The lamp should be connected across the transformer primary. As a wise old electrician once told me, "It's better to have a ground rather than to BE one."

Al Forbes
Gastonia, N.C.

PVC PIPE

I have been actively involved in audio at the home level for 40+ years, although I was unfamiliar with your publication until I read the *SB* 6/00 issue. My interests are as varied—both wide and narrow in scope—as *SB* appears to be. I especially enjoyed your piece regarding Yoshio Satake and his "museum" ("Showcase," p. 50).

This indicates that your magazine is capable of an element missing in the other "rags": namely, humor. So many audiophiles and wannabes take themselves way too seriously...even Lisa Astor in *Stereophile*, who, with tongue in cheek, is rather the Martha Stewart of the genre.

Does anyone fool around with HDPE sewer pipe, available in quite a few diameters, @ $1,200 for a 40′ section of 8″, for instance. This, a pretty inert material, and hard to work with in some ways, offers many advantages and opportunities for tweakers. For example, a pair of 6′ × 8″ sections sit on my listening room floor. They are quite tunable by way of a piece of plywood or potted plant on top, and varying amounts of glass, foam, or crumpled paper inside. It is only available, as far as I know, in basic black, but I fool 'em and cover with a sock of sky-blue silk.

John Thomas
Nome, Alaska

See Scott Wolf's 'A Pipe and Ribbon Odyssey," in SB 3/91, pp. 28–30. —Eds.

ELECTRONIC SPEED CONTROL UPDATE

Since my article "An Electronic Speed Control" (*TAA* 1/86) was reprinted in *The LP Is Back!*, I have received letters and telephone calls from several readers noting that the ILP power amplifier module and toroidal transformers are no longer available.

Fortunately, I have found substitutes for all of these parts. Plitron Manufacturing (www.plitron.com; e-mail: sales@plitron.com) has suitable transformers, and they are now a distributor for the Amplimo line of power amplifier modules, which are manufactured in The Netherlands. They can also be contacted at 1-800-PLITRON (1-800-754-8766) or (416) 667-9914, ext. 236 (Voice); or (416) 667-8928 (FAX). Ask for Helen Chen, customer service.

You should change the ILP HY60 power amplifier to an Amplimo A60. Note that Amplimo's model numbers refer to the 4Ω power rating. This module is rated at 40W into 8Ω, a bit higher than the 30W HY60. I do not recommend using the Amplimo A30, since it is rated at only 25W into 8Ω, and probably won't produce sufficient output voltage under load at the secondary of the step-up transformer.

All parts for the A60 power supply are also available from Plitron, which recommends a 037016201, 80VA power transformer, and the DE capacitor/rectifier assembly. Note that these power-supply parts replace the old ILP PSU-410 supply, which included a 4A027 power transformer and the PCB-410 circuit board with rectifier diodes and filter capacitors. Amplimo also recommends that the LRZ output muting relay be used with their power amp modules. The muting relay may be optional in this application, but it is so inexpensive that I recommend using it.

A schematic showing the implementation of the muting relay is available on the Plitron website in an applications note for the amplifier modules. You may also ask for a copy of this when you place your order. The step-up transformer was originally an ILP 1A010, a 30VA unit with dual 6V secondary windings, which are used in series to form a 12V *primary* in this application. The replacement is the Plitron 017010201.

The XR2206 function generator chip is still available from Jameco Electronics (Jameco 34972; www.jameco.com). The correct Bourns number for P5, the 10-turn 20k pot, is 3541H-1-203 (Newark 12F5004; www.newark.com). The Bourns H-491-3 counting dial is also available (Newark 12F9669). Newark notes that this counting dial is available only in the United States. Outside the US, try Bourns H-492-3 (Newark #12F9670). This is the same part, but with a break included (so the pot can be locked into one position).

Gary Galo
Potsdam, N.Y.

IT PAYS TO ADVERTISE

I really enjoy reading the editorials in *GA*. They're always insightful and informative.

The one in *GA* 2/99 was right on the mark, stating that parts suppliers were not advertising enough. I agree. However, I would go further, claiming that tubed-equipment manufacturers also do not advertise enough.

It seems as though every day when I open my morning paper I see pages full of hi-tech (imported) audio gear that sounds boring and looks cheap and ugly. I have yet to see even one piece of audio equipment that employs tubes.

If import audio is the norm, why not include Chinese tube amps, such as those appearing in the *World Tube Directory*? Could it be that the big shots who own the audio store chains fear that consumer confidence in the widely available hi-tech audio gear just might drop when audio enthusiasts see this hi-tech trash for what it really is? Would including tubed gear, in short, be "technologically incorrect"?

The tube-equipment manufacturers should catch the wave and *really* go public with their marvelous innovations. Not doing this is comparable to Mike Tyson fearing a shoe salesman. If the tubed-equipment manufacturers seek more audiophiles to road-test their glass, they must first let them know that it even exists. Once they show the mainstream audiophiles what they have to offer, they will surely reap their long-awaited rewards and be the giants they were some 30 years ago.

Neal A. Haight
Castro Valley, Calif.

RECTIFIED TUBES

I doubt that it is wise to use a 6×4 rectifier as shown in "Rectifier Tubes as Soft-Start Controls" (*GA* 3/00, p. 70). As mentioned in the RCA manual, the heater needs 6.3V-0.6A, which is two times the current mentioned. The maximum positive cathode to heater voltage is 450V, so even at a positive supply of 400V there is no risk of a breakdown. For a higher voltage of, say, +460V you can connect all filaments to, say, +60V.

As the high voltage builds up slowly during warm-up, there is no risk of an overvoltage when switching on the amp. But, in this configuration, capacitor C1 will immediately charge up to a high negative voltage. This greatly surpasses the specified maximum negative cathode to heater voltage of 100V and will destroy the tube and maybe more.

A.J. van Doorn
Amersfoort, The Netherlands

WORD USE

I am the electronics technician at the COSI Museum of Science and Industry at Toledo, Ohio. Last week, I was discussing with my supervisor and a friend an exhibit designed to demonstrate that W = EI. My friend, a network engineer, wished to set up a parallel exhibit in fluidics "to show that voltage is like the diameter of the pipe, and current is like the height." [Funny how computers have almost nothing to do with electronics!]

Of course, I explained that the pipe diameter was analogous to resistance, but when I got to height, or "head," I had a problem. I explained that voltage lacks the parameter "area," and that the analogy with pressure was therefore inexact, but close enough for most people. That was an evasion.

First of all, head is force/unit area. Since it is normalized to unit area, it is essentially a unit of force. If I had used the correct terminology, I would have spoken in terms of electromotive force, and the conflict would have vanished, but people understand voltage more readily than EMF. Thus, the price for avoiding Mr. Dell's "pressure of a voltage difference" is more confusion (*SB* Mailbox, "More Words," *SB* 5/99, p. 44).

I solved the problem in my exhibit by making two parallel statements: "Electric Force × Electric Current = Power," and "Volts × Amperes = Watts," thus separating usage from measurement. This is the case in Europe, I believe, where the symbols V and u are both used for "voltage." They would say, for example, "a 12V power supply which measured 11.93u." Unfortunately, we Americans [well, "Norteamericanos"] are not that far advanced.

As for the differences between voltage and "meterage"—I don't think your distinction holds up. Languages are full of examples of inconsistent patterns of use. Consider the series: Asians, Canadians, Africans, Germans, French, Finnish, and Japanese. And it doesn't get any better if we substitute "Frenchmen," "Deutchlanders," and so on.

We have a mixed bag in electronics. We have voltage, amperage, and wattage, but we also have the 'ances—resistance, susceptible impedance, reactance, and so on. (All of these are French suffixes. Hmmm.) So we actually have two distinct patterns. Another fact I find curious is that while "voltage" is a simple force often assigned to quality of a rate, "amperage" is a rate dressed up as a simple quantity. Go figure!

Actually, charge is the first quality in electronics (and is singularly free of suffixes, or even a clear plural form). All other qualities are derived from charge and cannot exist without invoking it. Next come quantity (Coulomb) and difference (voltage); then the ratios start, current, conductance...I absolutely refuse to go into the mechanics of charge! Chemistry ends with the atom, and electricity ends with charge.

Bob McIntyre
sideshowbobmac@thesimpsons.com

BUYER BEWARE

I have an assortment of used 12AU7/T7/X7s and noticed something that took me aback. Every one of my Mullard and Telefunken tubes have a defect in the heaters. When inserted into circuits with a 6.3V AC heater source, the heaters go table-lamp-bright when I switch on the equipment. The surge in brightness lasts for a second or two. I didn't pay much attention to it until a couple burned out.

The strange thing is that this occurs in all tubes made in either Great Britain, Germany, Holland, or France. For example, if I pick out two Sylvania 12AX7s, one made in Europe and the other made in the U.S., the European tube will act this way while the U.S.-made tube behaves normally. In DC heater-sourced circuits, the Euro tubes work well, performing up to their reputation.

If you're ready to re-tube a vintage integrated amplifier, put the Euros in the phono preamp stage, and U.S. tubes in AF amp and phase-inverter stages. Of course, certain amps have an AC heater source throughout the amp, so it may be best to refer to the schematic, or consult an expert, to determine which locations are AC or DC heater-sourced. Back when Euro tubes cost four or five dollars, it was nothing to really lose sleep over, but today they go for big bucks—a major supplier is selling Mullard 12AX7s for $60 a piece. I believe the defect takes place when the tubes become older and have been used for quite a few hours.

Japanese 12AX7s are as reliable, heater-wise, as U.S.-made NOS tubes. I hope Chinese tubes are at least equally as good as the Japanese tubes.

One brand of tubes I always see advertised in *Glass Audio* are J-J tubes. As I understand, these tubes are made in Slovakia. They look rather nice in the advertisement photo.

Whenever *Vacuum Tube Valley* compares tube types in their "shootout" articles, J-J tubes aren't ever mentioned. I have tried to obtain information on several occasions, but J-J hasn't responded so far. I would like to find out how they perform, which is difficult.

In conclusion, I'd say that U.S.-made tubes are a real bargain and should be the first choice for tube buyers.

Neal A. Haight
Castro Valley, Calif.

DRIVER CIRCUIT FOR 6AS7 AMP

Thank you for the recent articles[1-3] on low-mu triode push-pull (pp) power amplifiers. I found them so interesting that I have built one myself. However, the driver and input circuits are different, and I believe some readers who have embarked on this project may find the following information useful.

Low-mu power triodes usually require very large drive signals, probably in the order of 200V peak-to-peak. This may be two or three times higher than that used in a typical pentode or tetrode pp amplifier. To achieve this requirement, Stewart[1,2] uses a form of positive feedback, while Cottrell uses an interstage transformer. I prefer not to use positive feedback*, and I do not have an interstage transformer. Therefore, I set out to design an input/ driver circuit that can provide 200V peak-to-peak with "reasonable" distortions and gain.

The required devices and components must be readily available and cheap. The result is shown in *Fig. 3.* The complete design consists of two cathode-coupled stages, with the second directly coupled to the first. The first stage utilizes the most popular triode on earth, and the second stage uses the 5687, which is popular in many Asian designs. Simulations showed that the 5687 is more linear than the 12AU7 or the 6SN7 under this operating condition.

The output stage is not shown because it is very similar to those described in references 1–4, except that I use paralleled 6AS7s or a pair of 6C33C-Bs. Measurements for this circuit are shown in *Table 2* with some additional comments. This has been successfully used to drive either 8 × ½6AS7s or 2 × 6C33C-Bs in push-pull mode. I have not implemented the drivers of references 1 and 3 and cannot make direct comparisons. I would like to hear comments from others.

Using 6C33C-Bs instead of 6AS7s may be another alternative that readers can try and comment on. I tend to prefer the 6C33C-Bs. Reference 4 gives a good description on the use of these very powerful Russian triodes.

REFERENCES

1. J. Stewart, "A Different Triode Power Amp," *GA* 2/99, p. 50.
2. J. Stewart, "An Updated Triode Power Amp," *GA* 4/99, p. 46.
3. M. J. Cottrell, "6AS7 Amplifier," *GA* 6/99, p. 20.
4. D. J. F. David and J. B. Fortias, "Triode Power Amplifier 6C33C-B," *GA* 2/99, p. 1.

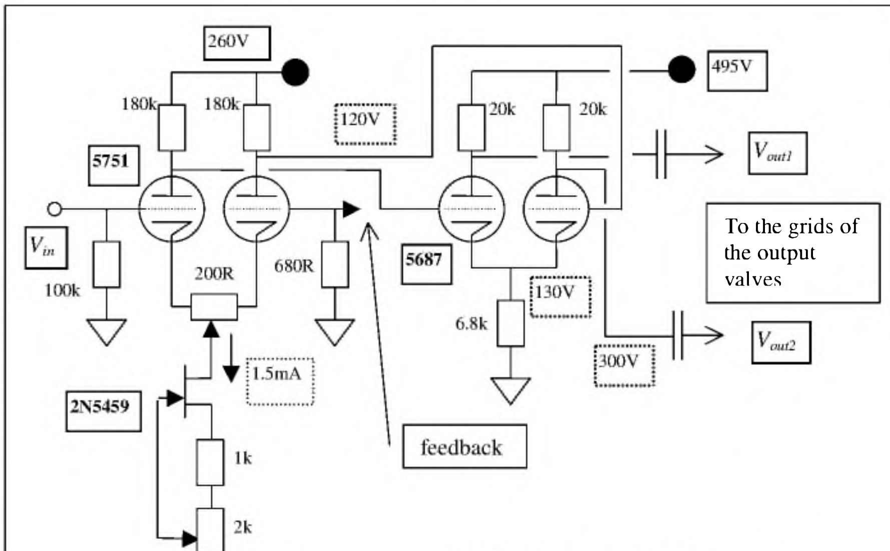


FIGURE 3: 6AS7 amp driver circuit.

G-Wong-3

TABLE 2

OPEN-LOOP CHARACTERISTICS

- Gain ≈ 300 (50dB)
- V_{out} ≈ 70V RMS @ around 1% THD (50Hz to 15kHz)
- V_{out} ≈ 80V RMS before clipping
- Frequency response ≈ 40kHz, –3dB

NOTE:

- Voltages indicated are not exact and may vary slightly.
- The main adjustment is to vary the two VRs of the first stage such that the two output voltages are identical (or as close as possible) and as symmetrical as possible–this can be easily done using a two-trace CRO.
- The 1k resistor is used to measure the bias current.
- Using well-matched input valves may eliminate the 200R VR and will extend the frequency response and increase the overall gain.
- 12AX7 may replace 5751 in the front stage.
- The overall open-loop gain of the complete amp is around 36dB (this is dependent on the output transformer used) and is ready to take a negative feedback of up to around 12dB. I believe 6–10dB will suffice.

* I believe the proper use of positive feedback will work, but there are other problems in addition to stability that I would like to avoid.

CC Wong
Dept. of Communication & Electronic Eng.
RMIT University
Melbourne, Australia

John L. Stewart responds:

It is gratifying to know that someone is reading your stuff and then trying something new. As with all designs there are tradeoffs. My objectives in regard to the "Different Triode Power Amp" were to sidestep some of the more serious problems associated with higher power triode amplifiers.

I intended the design to:

1) Avoid dangerous high-voltage power supplies
2) Use readily available, inexpensive components
3) Eliminate frills such as driver transformers, DC heater supplies, and so on

At this point, I would like to propose another class of amplifier as follows:

CLASS USA
U—understandable
S—safe
A—affordable

With regard to CC Wong's amplifier, I have the following observations.

The first stage using the 5751 (or 12AX7) may have problems when feedback is connected to close the loop. Using a semiconductor as a current source is a good idea. However, in the proposed circuit there is not enough common mode tolerance. A 9V battery is being used here; 4.5V are used in passing through the biasing resistors. That leaves only 4.5V to operate the 2N5459 and provide current to the 5751 differential pair.*

Using my development amplifier, I connected feedback to give an overall gain of 6 from the input to the 8Ω output terminals. I measured 1.62V RMS at the cathodes of the first differential pair while the amplifier was delivering 20W. That would be 4.58V peak-to-peak. If the 2N5459 bias circuit were used, there would be lots of distortion. I would use another 9V battery to get 18V and a margin of safety. The development amplifier avoids this problem by using a −150V supply.

The 5687 is a very good twin triode and can easily deliver large output as a driver to a power stage. The 7119 is another twin triode that could serve equally well.

In my library I have data covering the 5687 but no curves. In order to make a comparison with the 6SN7GTB I had used in the development amplifier, I needed, in particular, a plot of the zero bias curve to overlay this onto the 6SN7GTB curves and from that determine how much more voltage swing to expect from the 5687.

To get that information I haywired a simple circuit with suitable load resistors (Fig. 4). *Power was supplied by a variable regulated power supply set to 365V. The load resistors measured 21.6k. These conditions are a close match to that of CC Wong's circuit. I tried four 5687s. Voltage measured at the test points for the eight triodes was from 48 to 51. If anyone tries this, the load resistors should be 10W.*

Using the 3/2's power law, I plotted the zero bias line for the 5687 onto the plate characteristics for the 6SN7GTB family (Fig. 5). *You may notice the graph is labeled 6J5. The 6SN7* family is actually two 6J5s in the same bottle. Scaling on this graph was more convenient than that given for the 6SN7 family.

The 5687 can turn on about 63V farther than the 6SN7 with a 20k loadline. Needless to say, I can't go any farther the other way. If you assume cutoff to be at 350V, the 5687 can swing 300V. The 6SN7 will be good for 235V. The 5687 gives a 28% improvement. These numbers are rather ideal since they assume a DC load line. In practice, both will be reduced by rotation clockwise of the loadline under AC load conditions.

Both swings are extended 100% in the development amplifier by bootstrapping, which acts like a tracking power supply that makes available extra voltage when required. For those who are concerned that it is positive feedback, it measured 1.38dB in the development amplifier.

Total DC across the driver stage in CC Wong's amplifier is 495V. He will need an extra power supply either separate or piggybacked to the 260V supply. Either way would require an extra HV winding on the power transformer. I managed to avoid that complication by using a negative supply for part of the power requirements. It uses the same winding on the transformer as the positive supply. He may also need an extra heater supply to avoid heater cathode insulation problems.

*Common mode tolerance—ability of the amplifier to respond when both inputs are driven in the same direction. This is limited by the supply voltages.

Matthew Cottrell responds:

CC Wong raises some interesting points here, to which I would add a few comments, as follows:

1. Yes, low-μ power triodes require large drive signal amplitudes. In my example, a near-copy of McProud's amplifier, the interstage transformer provides about 190V peak-to-peak grid voltage to the 6AS7 output tubes. When I first proposed this article to Ed Dell, his initial comment was along the lines of "...I will be interested to see how you drive the somewhat difficult 6AS7 output stage...." As I remarked in my article, I only built this monstrosity because I had happened upon some interstage transformers. I probably wouldn't build it again.

2. While quality interstage transformers are fine, classic solutions to high drive requirements, they are very difficult to source nowadays, so using an active solution is a really good idea. In my opinion, the differential amplifier—and other cathode-coupled variants—shown by Wong is perhaps the best alternative.

The differential amplifier is, I believe, sadly under-utilized in audio in general. It can serve as a gain stage with really good common-mode (and power supply) rejection, as a phase splitter, as a balanced amplifier, as a null indicator, and so forth. It has good gain and impedance characteristics, and can be improved even further by the addition of a constant-current source as the cathode resistor, as Wong does in his first stage.

In addition, one of its best features is that it provides a second input at ground potential for the application of feedback, as Wong shows. This eliminates one more capacitor, as compared to applying feedback to the cathode of a single-ended input stage, as is frequently done.

3. The 5687 appears to be an acceptable dual triode for this application, although I am unfamiliar with it in practice. I still prefer the 6SL7 and 6SN7 for almost all such applications (I just don't like miniature tubes). If Wong has proven to himself that the 5687 is a more linear device, then I stand corrected.

Again, though, regardless of the tubes employed, I think Wong's circuit is a marvelous way of developing the high drive voltage required for the low-μ output stage, be it 6AS7s, 6C33C-Bs, or other choices. I will probably breadboard his circuit to try it myself!

4. In general, I agree with Wong that the use of positive

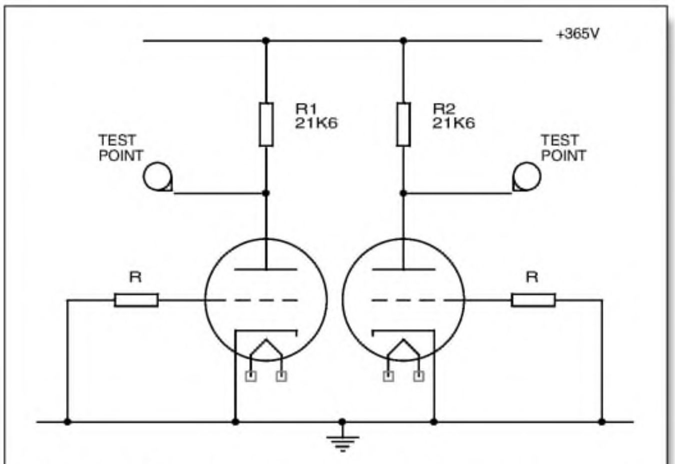


FIGURE 4: Sample circuit used to obtain 5687 data. G-Stewa-4

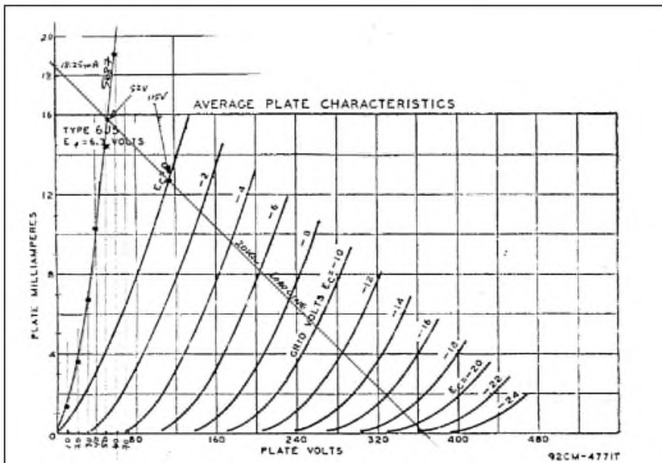


FIGURE 5: 5687/6SN7GTB comparison. G-Stewa-5

feedback is difficult to implement at best—I have built far too many [inadvertent] oscillators in my time, and don't care to tempt fate by deliberately introducing positive feedback into an otherwise stable circuit.

CIRCLOTRON

My notes in *GA* 4/00 on selection of the load impedance for the 6AS7/6080 bootstrapped amplifier when compared to Monny Nisel's amplifier are incomplete (Letters, p. 62). I somehow managed to overlook that his amplifier uses the Circlotron circuit (*Fig. 6*).

The Circlotron has an important advantage when compared to an ordinary push-pull amplifier. The output tubes are driving the load in parallel rather than series as they do in common push-pull. That means the output transformer required to properly match to the load needs to be just one-quarter of the impedance of that used in the common push-pull configuration. His amplifier using a 1kΩ load impedance translates to a 4kΩ load impedance if it were in the common push-pull hookup. The advantage conferred here is that it is easier to build a good output transformer as you lower the impedance.

Since nothing comes free, there is as well an important disadvantage. The Circlotron needs two identical high-voltage supplies. Also, the amplifier output must drive the stray capacitance of the power supply. Luckily, output is taken from the cathodes, which are a low impedance drive point. With enough power triodes, you can drive a loudspeaker directly without an output transformer.

Power transformers with the required two HV windings and output transformers with matching impedances of one-quarter that commonly found have never been easily available as standard catalog items. As a result, few amateurs or experimenters have tried this kind of circuit. That includes yours truly. Perhaps next year?

John L. Stewart
King City, ON
Canada

HELP WANTED

I have four ribbon horn tweeters from Decca, or more precisely, the "Decca London Horns." Where can I find replacement ribbons? Money is not an issue.

Harry Timmerman
HTI@joco.nl

Several years ago, we published an article by a man whose hobby is rebuilding microphones—including ribbons. He had a method of finding industrial-type film, aluminum sputtered, which he formed in a jig and installed in ribbon mikes. I don't

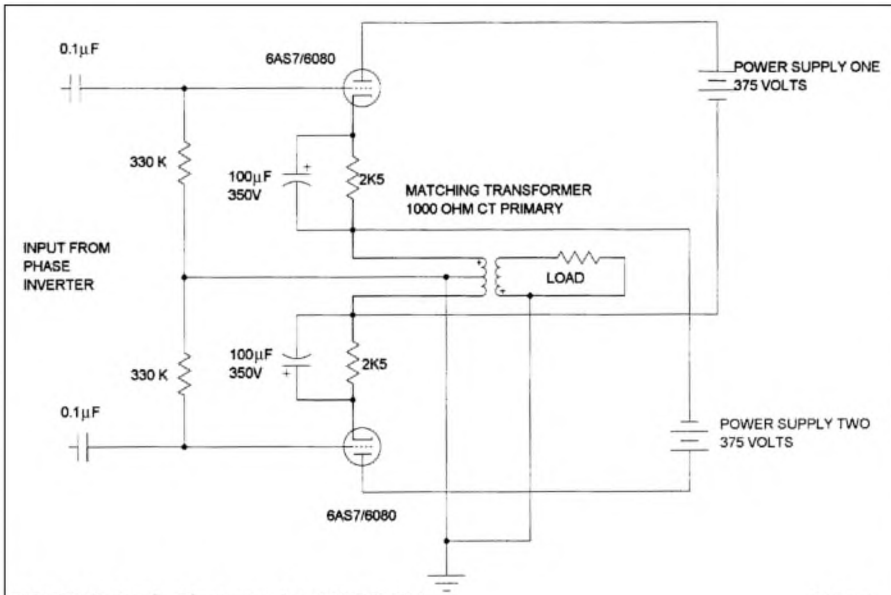


FIGURE 6: Basic Circlotron for 6AS7/6080.

know whether his current address is valid or whether he can help, but you might try him and learn to make replacements yourself. His address is: David Royer, 628 W. Amerige Ave., Fullerton, CA 92632. —E.T.D.

I have acquired a valve TV and need a schematic. I cannot find anyone who can help. The set is an HMV Boston Model PP-AE.

Peter Laurence
laurence@intercoast.com.au

I have a tube preamp made by GSI musical electronics model 5tp. In my search, I have been informed that the company no longer exists. Where can I locate a schematic or any available information on this unit?

Scott Cromwell
kronell@wans.net

Where can I find the Quad Cap for the Dynaco MkIII mono amp? Is there an acceptable replacement? I'm also looking for a diagram on how to add a second bias adjustment to individually adjust the tubes.

D. Sherfy
thx1326@swbell.net

I am looking for someone—preferably in California—who can modify my Dynaco ST-70 per Norman L. Koren's article in *GA* 1/92 ("Modifying the Dynaco ST-10 with Triode Mode").

Alberto Sarmiento
2837 Galena Ave.
Simi Valley, CA 93065

I'm searching for a cabinet builder for a vintage pair of Tannoy 12″ dual concentrics. Might you be able to recommend someone?

D. Milch
dmilch@bellatlantic.net

Which of the various types of phase splitter circuits is the one also known as a "Loyez" phase splitter? This seems to be a colloquial term for one of the types that uses a double, or two single, triodes. If it is not a commonly known type, then I would be most interested to see a basic sketch of its circuit, as may other readers.

Chris Logan
1/14 Brodie Street
Paddington
N.S.W. 2021
Australia

I am looking for PSpice models on valves, especially small-signal types such as ECC81/82/88 which I can import into Microsim 8 in order to simulate in a hybrid design of my own. I have managed to produce the symbol and the template within the software, but now need the subcircuit information to go with it. I have found several relevant references to previous articles in *GA* on the Web, so I know there must be something out there. ❖

Paul Bailey
Paul.Bailey@theseed.net

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

Book Reviews

GEC Audio Tube Data

Reviewed by Larry Lisle

GEC Audio Tube Data: Data and Amplifier Designs KT66, KT77, & KT88, Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, (603) 924-9464, FAX (603)924-9467, e-mail custserv@audioXpress.com, $16.95. BKAA46

It wasn't that long ago, back in the "golden age," that the only tube manuals most people knew about were the RCA and the GE. They were the ones sold or given away by your friendly neighborhood tube and electronic parts distributor (Remember them?). If a tube wasn't in one of those manuals, we never thought about using it, which was why RCA and GE made their manuals so widely available in the first place!

Well here's at least one way in which the present "silver age" is superior: There's a lot of tube data readily available, thanks to the efforts of the publishers of reprint editions. Of course this material was around in the old days, but it didn't do us any good if we never heard of it.

An excellent example of a reprinted tube data book is *GEC Audio Tube Data.* It begins with a preamble, written in 1979, comparing audio valves to semiconductors. I knew some of this information, but there was some I had never read before.

EUROPEAN STYLE

The book continues with specifications for the KT66. If you're not familiar with European tube data sheets, you're in for a surprise. The American manuals just don't compare. For example, not only is total distortion provided under various operating conditions, but intermodulation distortion information is listed as well—and the method is specified. No less than eight sets of curves are included for just one tube type.

GEC Audio Tube Data continues with several complete amplifier circuits for the KT66. It provides complete data, curves, and parts lists for the circuits as well! If you don't happen to have a couple of KT66s on hand, the book offers plenty of ideas that can be used with other tubes.

The next section follows the same format in 38 pages of data for that wonderful tube, the KT88. I've long been a fan of the 6L6 and its relatives when it comes to beam tetrodes, but to think that no one could improve upon them in the years after their introduction (in 1936) isn't a good bet. The KT88 is still in production and probably will be available for a long time.

There is too much information in this book to cover in detail, so I'll just mention a cute little 30W ultralinear amplifier that produces only 1% distortion with no feedback. A complete description of the circuit is given, along with a parts list.

One aim of the European tube books is for the hobbyist to build—and be happy with—circuits using the manufacturers' tubes. The circuits tend to be easy to make without much in the way of test equipment and without hard-to-find components.

I DIDN'T KNOW THAT

A number of other circuits are also described, including a 400W giant that uses multiple pairs of output tubes. Interestingly, the book points out that, with increasing pairs of output tubes, it becomes easier to balance the current between the halves of the push-pull circuit. I never thought of that.

These days, there's a lot of interest, especially among amateur radio operators, in "old fashioned" AM transmitters and good audio. This book might be exactly what they're looking for.

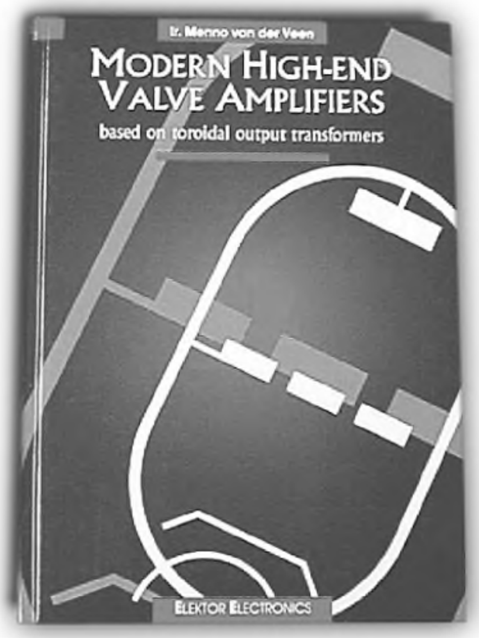
The section closes with an appendix on intermodulation in ultralinear amplifiers. Did you know that the triode connection doesn't give the lowest intermods, except below 10W? I didn't. I haven't done anything with ultralinear amps in a while but I'm going to have to look into this.

The section dealing with the KT88 is followed with similar information on the characteristics of the KT77. Following this, though, is information on the PX4 and the PX25 triodes. Information on these tubes is just not available in a lot of places! In fact, information on any of the tubes mentioned is certainly not common.

Sooner or later, this little book will also go out of print. You might not be interested in building a push-pull KT88 or a single-ended PX25 amplifier now, but you might be kicking yourself later on if you don't pick up a copy of *GEC Audio Tube Data* while you can. ❖

Modern High-End Valve Amplifiers

Reviewed by Larry Lisle



Another very interesting modern book about tube amplifiers has just been published. It's *Modern High-End Valve Amplifiers* by Menno van der Veen of the Netherlands, (available from Old Colony Sound Laboratory, PO Box 876, Peterborough, NH 03458, 603-924-9464, FAX 603-924-9467, custserv@audioXpress.com, $59.95).

This book has a different slant than many tube amp books, emphasizing the use of a toroidal output transformer, but most of the material applies to conventional IE core transformers as well. The wealth of material here about transformers is not available elsewhere in such a readable format. There's also a lot of math, but unless you choose to design and build your own transformers, you can skip these parts and still understand the important points of each chapter. The book also contains many circuits that you can simply copy or analyze from the accompanying text.

In this review I'll concentrate on the first part of the book, the theory part, since I think it will be of the most interest. The second part puts the theory into practice in a number of very interesting amplifiers for a variety of power levels and purposes.

EARLY CHAPTERS

The book begins with a couple of pages about the author, who says he has written 360 articles for Dutch audio magazines, and whose work has appeared in many American publications, including *Glass Audio*. The introduction outlines the organization of the book and states its purpose as a "textbook as well as a do-it-yourself guide."

Chapter 1, titled "Why Valve Amplifiers?" begins with a comparison of tube and transistor amplifiers and comes down strongly on the side of tubes. This is followed by a brief section that answers the question, "How do valves work?"—covering about four pages and cutting very neatly to the heart of tube amplification. The author then goes on to discuss various types of valves, why an output transformer is needed, and the topology of a basic push-pull and basic single-ended amplifiers, and closes with a bibliography of books and articles relevant to the chapter. A complete bibliography is also found in the appendix.

In Chapter 2, which discusses output transformer specifications, the author does an excellent job of making a complex subject quite understandable and reducing the specifications sheet to the following: the turns ratio, the primary inductance, the primary leakage inductance, the effective internal capacitance of the primary, the total resistance of the primary winding, and the total resistance of the secondary windings. He closes with spec sheets of transformers he has designed, which will be used in circuits in later chapters.

Chapter 3 starts with a description of the variable nature of loudspeaker impedance. The author continues with a discussion of triode, pentode, and ultralinear configurations and the effect of these on plate resistance. He then explains the damping factor in a very clear and concise fashion. If you've wondered about damping factor but been afraid to ask, the answers are here. The chapter concludes with a discussion of the -3dB high-frequency limit. The author shows it's dependent on the tube configuration (triode, ultralinear, pentode) and speaker impedance.

MATH SKILLS

Chapter 4, "The Output Transformer in the Complex Domain," is just as complex as it sounds, but van der Veen does an excellent job of making the material usable by the average DIYer. He starts by showing the current through a coil, such as a transformer primary, produces an opposing current. He then explains real and imaginary components at different frequencies, phase angles, and different phase distortions.

A phase difference in the sound coming from a speaker system that's the same at all frequencies is of no concern (we're not talking feedback here); it's when different phase responses occur at different frequencies that problems arise. He says that, with a good transformer, this won't happen until well above the audio range, and you won't notice any degradation of transient behavior. However, in the real world there are some limitations: with pentodes, errors may just creep in at the upper end of the audio range; in the ultralinear mode it is still possible to stay in the error-free domain if you use the proper coupling; the best behavior is obtained with triodes or triode-configured tubes. I've never read a better presentation of this material than the author provides here.

For light reading, turn to Chapter 5 for "Frequency-Domain Calculations for Toroidal Output Transformers." The author really gets into the math of transformers. Although he uses only algebra, the equations become quite complex, and he recommends the use of a computer for complete solutions. He also suggests some simplifications that will give good approximations. It's well to remember that you don't always need to solve equations to get information from them. Just knowing that increasing "this" causes "that" to go down, for example, is often all you need to know.

Chapter 6, "Theory of Overall Negative Feedback," is full of information, but, again, there's some serious math involved. Here's a sample of what you can calculate: How the phase splitter and preamp can significantly restrict the frequency range to less than that obtainable when only the output tubes and transformer are considered; how a given amount of negative feedback changes the transfer function; how stabilization in the frequency domain prevents ultrasonic oscillation; and how series grid resistors really work.

FINAL CHAPTERS

Chapter 7, "Output Transformer Low-Frequency Tuning," considers the low-frequency behavior of a tube amplifier. This is an important chapter. Although there's some math, you don't need to solve the equations to understand the main points.

Did you know that if you use a transformer with a very large primary inductance, the bass actually might sound weaker? This is because there's less distortion in the bass with a large transformer, and bass distortion has a different effect on our ear/brain system than does mid- and high-frequency distortion. The author leaves it to you to decide whether extra harmonics of low bass tones are desirable or not, but points out that you should never use oversized transformers with guitar amplifiers because the warm sound character produced by a transformer with a small primary inductance is a major factor in producing the specific guitar sound. If you like bass, you'll need to read this.

In Chapter 8 the author describes "Special Output Coupling Techniques" and deals with special transformers, such as those with cathode feedback windings, separate screen grid windings, and unity-coupled transformers, and how to use them.

Chapter 9, "Single-Ended Toroidal Output Transformers," will be of great interest to many readers, but contains a major gaffe in a circuit diagram: a 300B is shown with a cathode. I'm sure the author is aware that a 300B contains only a filament, grid, and plate, so perhaps he uses the cathode symbol as a kind of shorthand or stylized way of drawing a schematic (the cathode resistor and capacitor being connected to an imaginary cathode for clarity, instead of to the center tap of a transformer or resistor across the filament). Maybe it's just a gaffe that snuck in while the book was in production and nobody caught it.

In Chapter 9, the author designs an amplifier output stage using a 300B. If you're planning to build such an amp, this material can be very useful, since everything is presented clearly. "With the calculation tools given here, you can choose any combination (of parameters) you like." He then discusses the design of a transformer for single-ended output stages and says the factors obey the law of "conservation of aggravation" because most of the factors represent diametrically opposed requirements.

The author then extensively discusses low-frequency behavior, high-frequency behavior, high-frequency behavior and fine-tuning, insertion loss, maximum allowable primary direct current, and, finally, phase distortion and differential phase distortion. This is well worth reading if you're planning any single-ended amplifier projects.

The rest of the book is devoted to building practical amplifiers; including both high-end and guitar applications. There is a wealth of information on practical design and construction that will be of benefit to anyone building amplifiers. This is an excellent book and will be useful whether you are considering using toroidal transformers or plan to stick with conventional models. ❖

Audio Aid

Improving DVD & Video Performance

I recently bought a budget Sony DVD player, connecting the sound directly to my stereo preamp. In an effort to obtain all that the DVD player could offer, I did the following:

1. Applied lead foil to the tops of the RF shielding cans covering portions of the DVD player circuitry and adhered a damping layer inside the metal cabinet. This removed vibration-induced hash from high-frequency audio.
2. Reconverged the tube of the 27″ Zenith color TV to improve sharpness.
3. Studied the schematic for the TV set—I have always received excellent treatment from the Zenith service personnel, who offer very good literature at fair cost. This material also documents the adjustment of the gun cutoff voltages and gains, using the remote in the service mode. This welcome resource is thoroughly described in the manual.

The picture quality of my Sony computer monitor leaves commercial TV in the dust, so, to even up the situation, it seemed that a little tweaking in the TV set was in order. *Editor's note: Use extreme caution in working on a TV chassis! Make sure the unit has been unplugged for at least 4 hours.—Ed.*

It appears that Zenith uses surface-mount devices where possible, and electrolytic capacitors where large values are needed. Due to the well-documented faults of electrolytics, I decided to help them with film-capacitor bypasses. Zenith uses electrolytics in the following situations:

- To couple audio signals between devices and to the output jacks.
- To couple video signals between devices and to the high-voltage gun drivers on the picture tube.
- To provide AC grounds in amplification stages and filters.
- To filter the DC power-supply voltages.

I settled on paralleling small-value electrolytics with 0.027µF polyester caps, and using 0.47µF caps on the large electrolytics, to simplify my Digi-Key order. I used 300V caps for the high voltages at the gun drivers. I don't have the skill to remove and replace components from delicate foil traces, so I leave that option to more adept readers.

I drew up a list of subject caps from the schematic and removed two PC boards from the set. I carefully identified the electrolytics and soldered their helper caps onto the back of the boards, which I then reinstalled. So far, everything was fairly easy, if painstaking.

I carefully reconverged the picture tube. If you don't have a test-pattern generator, DVDs that serve the same purpose are available. I always end with converging one dot in the center of the picture, which I view with a 15× magnifying glass. This close, no white dot is seen, but rather a series of vertical bars of the three colors.

Adjust the 4-pole and 6-pole tabs (well-documented in your service literature) to line up the bars and give them equal height at the same locations in the dot (an ellipse, up this close). Tweak the focus control, then button up the set. Now you can adjust the usual options, such as tint, color intensity, and brightness, using the remote from the comfort of your couch.

Is this all worthwhile? For the audio, a very strong yes. Music and voice, even over the set's own speakers, are very enjoyable, free from hashy highs and spitty voices, and containing enough musical overtones to give a hint of high-end sonics.

For the video, a resounding yes. Although the Zenith is still not a computer monitor, stunning gains are present. More complete realization of the broadcast television standards makes for a delightful picture as viewed from a sofa. Picture sharpness is hugely improved, and colors are rendered with delicate shadings and hues. Details are clear and crisp, particularly on the black/white DVD version of "Casablanca," but also on broadcast program material, a big winner from these changes.

This has been a great adventure—easy and inexpensive to achieve—and has resulted in picture and sound quality only surpassed by very expensive monitors...and HDTV. ❖

Darcy E. Staggs
Orange, Calif.

New Chips on the Block

Analog Devices AD1885 Codec

By Charles Hansen

The Analog Devices AD1885 AC'97 encoder-decoder (codec) is bundled with third-party applications for the latest iteration of its v2.0 Sound-MAX motherboard sound system. The AD1885 provides stereo analog I/O on PC motherboard and peripheral devices as part of a high-quality PC audio signal chain. The AD1885 includes high-fidelity A/D, DAC, and sample-rate converters, as well as power amplifier and programmable gain blocks.

Technology from partners Sensaura and Staccato Systems delivers more realistic audio for Internet, voice-recognition, audio-synthesis, and 3D-gaming applications to the complete spectrum of PCs. Sensaura handles the 3D positional audio functions, employing a reverb-based model that requires less computational power than the reflection model. Staccato Systems supplies wavetable audio-synthesis that supports General MIDI (GM) and Downloadable Sounds (DS) standards.

The AC'97 features include variable sample rate audio, multiple codec configuration, and external audio power-down control. The AD1885 has greater than 90dB dynamic range, four analog line-level stereo inputs, a mono mike input with 20dB preamp, two analog line-level mono inputs for speakerphone and PC BEEP, a high-quality digital CD input with ground sense, stereo line-level outputs, and power-management support.

Enhanced features include full-duplex variable sample rates, supported from 7040Hz to 48kHz with 1Hz resolution. Up to three codecs can be chained on a single 5-wire interface. Jack sense pins provide automatic output switching. A mobile low-power mixer mode, extended 6-bit master and headphone volume controls, digital audio mixer mode, and Phat™ Stereo 3D stereo enhancement are also included. ❖

Analog Devices: www.analog.com or 781-329-4700, Norwood, Mass.
Sensaura Ltd: www.sensaura.com or +44-20-8848-6636, Middlesex, UK
Staccato Systems: www.staccato.com or 650-254-1971, Mountain View, Calif.

AKM Semiconductor AK4101, AK4102, and AK4103

By Charles Hansen

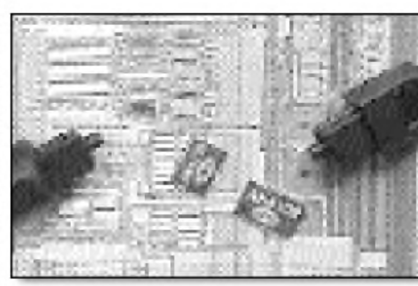
AKM Semiconductor (www.akm.com) announced three digital-audio transmitters to support the emerging 192kHz digital-audio sampling rates. The chips support standard formats such as the consumer audio S/PDIF and pro audio AES/EBU.

The AK4101 8-channel transmitter has four RS-422 differential drivers integrated on a single chip. This device is ideal for digital mixers, set-top boxes, surround-sound applications, and effects processors with multiple digital outputs. The small 44-lead LQFP package allows for a very space-efficient board layout. A common clocking scheme and programming interface reduces the component count.

The AK4102 4-channel transmitter provides two output channel pairs, an S/PDIF, and an AES/EBU output using only one device. The AK4103 offers a stereo transmitter in a 24-lead VSOP package.

The device family allows eight audio input formats, including IIS, Right Justified, Left Justified, and Master/Slave modes of operation with 16- to 24-bit word sizes. These devices, which are compatible with industry-standard audio receivers, can operate with a microprocessor or in stand-alone hard-wired mode.

Contact Richard Kulavik at 888-256-7364, or icinfo@akm.com for prices and engineering samples. ❖



Micronas MAS 3509D MP3 Decoder

By Charles Hansen

The successful introduction of the MP3 flash player midway through 1997 established a new digital-audio standard. It also marked the introduction of the Micronas MAS 35xx audio processor family. Micronas is now the world leader in MP3 decoder-ICs. Their MAS 3507D MP3 decoder powers many of the first-generation MP3 players.

Music titles compressed in MP3 format are downloaded from the Internet onto home PCs and, from there, transferred to a portable MP3 audio listening device. The MP3 player has an integrated or plug-in flash memory chip with a capacity of 32 or

64MB. This is sufficient for a playing time of 30 to 60 minutes. The compressed data is decoded and played back in real time.

The latest Micronas codec, the MAS 3509F, adds several capabilities. It integrates the D/A converter, headphone-output amplifier, S/PDIF I/O support, and clock-generator circuitry that previously required the separate DAC 3550A chip. A DC/DC converter provides dual supply voltages that you can independently configure within a 2.0 to 3.3V range. The converter includes a programmable-threshold, low-battery (<0.9V) warning indicator.

Unlike the earlier chip, the MAS 3509F also encodes analog mike and stereo-line inputs to a 16-bit A/D adaptive-differential PCM stream with 60dB signal-noise ratio, which Micronas calls FM-radio quality. (The quality-limiting Secure Digital Music Initiative, SDMI, prevents higher fidelity.)

The MAS 3509F runs up to 70MHz with a lower average power consumption of 65mW, and dynamically scales its RISC and DSP CPU core operating frequency to the incoming sample rate.

Micronas' "Perfect Audio" covers a set of various baseband features to enhance audio quality. Thus, the MAS 3509F is equipped with a virtualizer (pseudo-3D sound synthesis), an equalizer (parametric or 5-band graphic), dynamic bass boost, and automatic volume control.

Micronas supplies IC products and software for the key digital-audio coding processes and associated applications. *Table 1* lists the IC products that are currently available. You'll find more detailed information in the individual product descriptions at www.micronas.com.

In response to the continuing miniaturization of equipment, Micronas also supplies its digital-audio ICs in a particularly small housing (ball grid array, BGA).

The chief applications for Micronas' video and audio IC families are television receivers, covering a broad performance range:

MP3 devices—MP3 players with a flash memory (Smart Media Card, MMC, Compact Flash Card), Clik! (Iomega), or Microdrive, audio home servers.
MP3 integration in existing devices—e.g., CD players, mobile telephones, car radios, in-seat audio systems for cars, minivans, and public transport.
Voice recording and playback—language trainers and electronic dictionaries.❖

TABLE 1

AUDIO DECODERS	
MAS 3507D	MP2/3 decoder, voice codec
MAS 3509F	Single-chip MP3, AAC decoder, voice decoder, A/D and DAC integrated Dolby Digital decoder
MAS 3528E	Decoder for AC3 and MPEG-1 layer 2, A/D, DAC, power management G.729 codec
MAS 3504D	G.729 encoder/decoder
D/A CONVERTER	
DAC 3550A	D/A converter IC for data rates from 8–50kHz

Classifieds

VENDORS

High-End phono, line, poweramp, crossover kits, ladder attenuators, audiophile components. www.borbelyaudio.com

SERVICE/ALIGNMENT and operating manuals, $12.50; many original parts. Expert restoration/repair service. "***The Fisher Doctor",*** Al Pugliese, 27 Daleham St., Staten Island, NY 10308, (718) 948-7489, AOL: "FISHERDOC."

The LP track carries both stereo channels vs. alternating pits on the CD. The LP remains superior with "tight coupling for audio," $15; using an "OTL amplifier circuit," $35; driving efficient "Bass Horn" plans, $50. Visa, MC. ***STEEN AUDIO LAB***, PO Box 2185, Vancouver, BC V6B 3V7, CANADA.

AD INDEX

Classifieds
Continued

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Wool batting for audio insulation. Measures 18″wide × 3′long, 4-6″thick. $12/1lb, $23/2lb plus S&H. Contact (540) 254-6691 or rebecca@crescentmoonfibers.com

Tubed Altis D/A, latest modifications by Jerry Ozment. Warranty, $2,200. 610-291-5339.

AMPLIFIER SCHEMATICS: 7 class A schematics for $12 U.S. or 3 class AB schematics for $8, send payment by check to: Jean Tarzi, 5894 Grande-Allée, St-Hubert, P.Q. CANADA, J3Y-1B3

WANTED

TUBE HIFI, HORN SPEAKERS, and HIGH END. Altec, Marantz, JBL, Mcintosh, Electrovoice, Fisher, Scott, Dynaco, Leak, Eico, Audio Research, CJ, Quad, Bedini, etc. Sonny, Phone: 405-737-3312 Fax: 405-737-3355, sonnysound@aol.com.

YARDSALE

WANTED

Schematics of Ron Sutherland amplifiers, Audio Research LS5MKIII, LS16, and LS25, AR Reference Phono and PH1, Convergent Audio Technology SL 1, and Ultimate. Will pay for copying, handling, and postage. If you have the above or other interesting schematics of high-end equipment, please make an offer. Reinhard Hoffmann, Wiembecker Str. 1, 32657 Germany, FAX 05261 87586, e-mail: in.audio@t-online.de.

FOR SALE

NOS Tubes: RE-604, AD-1/Ed, EB-III, PX4/25, KT-66, KT-88, EL-34, RGN-2004/2504, and so on. Klangfilm poweramps KL408/502, KL402, KL32609 (RE-604PP + RE-304SE) KL-711 (ED/PP), Eq.: KL-002 (Neumann SEV2!), KL-RZ-057 (all phonocurves), Maihak V41 (best microphone tube pre amp ever), Telefunken V-74 tube microphone pre amp, Neumann MC-input transformers, Germany, FAX +49 6544-990715, e-mail: old.school@t-online.de.

Musical Fidelity X-24-K DAC—used 2 mos., $300; X-PSU power supply—used 2 mos., $150; X-CANS Headphone Amp, never used, $200. Shipping not included. All three for $600. All include original manuals, boxes, and accessories. Chris, (918) 663-9565 or HAL84CH@cs.com.

Tube amplifiers: Stromberg 21, 22, AU36, Bogen M120, Challenger. DuKane 1A360, 435, 475, S/S 803 with 5-mike mixer. School system, 120W AM/FM tape tone alert call back, 1931 Class-"A" final stage uses 845/866s DuKANE 13-mike mixer. Send SASE Edward Scribner, PO Box 279, Schoharie, NY 12157, (518) 295-8819.

A private collector sells a few thousand fine and rare tubes. Send $5 for catalog and postage to M. Hall, 62 Bombar Rd., Toronto ON M1C1C9, Canada, (416) 282-0358.

Yardsale guidelines

To publish your free personal ad in *audioXpress*, simply follow these guidelines:

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12. Noncompliance with any of these guidelines will result in your free personal-ad submission being refused for publication.

Glass Shard

Taming High Line Voltage

During my efforts to restore an old Harman Kardon stereo amplifier and a couple of old H.H. Scott mono integrated amps, I noticed that all of my voltage measurements were consistently 5–10% higher than in my schematics. This was a concern to me because those old power-supply capacitors are under plenty of stress. After a quick bit of detective work, I identified my home's line voltage as the culprit.

I monitored my line voltage for a week and discovered that it consistently ran at 121–123V AC. This is significantly higher (about 5%) than the 115–117V AC line voltages common in the 1950s. So I was determined to find a way to lower the line voltage supplied to my amps. (All of my schematics indicate that the measurements were taken with a line voltage of 117V AC.)

WIRING SOLUTION

My first thought was to use an auto transformer to drop the 123V AC to 117V AC; however, these are heavy, hard to find, and expensive. A Variac® is an alternative, but I certainly wasn't going to dedicate one to this application. The approach that I came up with is simple, elegant, less expensive, and much lighter.

My solution was to wire the output of a 6.3V AC 20A filament transformer in series with the line voltage, making certain that it was in reverse phase with the line voltage (*Fig. 1*). By wiring the transformer output in series and reverse phase, the line voltage is reduced by an amount equal to the filament transformer voltage.

Select the filament transformer output voltage and current rating to suit your needs. The primary winding of the filament transformer does not need to be rated for the full current load, just the secondary. (This cuts down the weight.) My transformer had a center tap so I could have used it to drop the voltage by 3.5V AC instead.

I added a line cord, a switch, and a GFCI receptacle to a small electrical breaker box to complete the project. The end result is a 6–7V AC reduction in the line voltage. My vintage amplifiers are now supplied with 115–116V AC, and I am sleeping much better.

CAUTIONS

The transformer should be of good quality and must be overrated for the application so that it doesn't become hot. All of the components in my voltage conditioner are rated for 20A with the exception of the line cord, which is rated for 13A. Although I don't expect to ever pull more than 10A from the supply, I will be replacing the cord with a 15A one.

Watch out for transformers that hum. My transformer has a very low hum that is barely audible. Encasing it in the heavy steel breaker box has virtually eliminated all traces of the noise.

The switch and the fuse or circuit breaker are optional; however, I recommend that you use a fuse if your filament transformer is rated for less than 15A since you can't expect your home's 15A circuit breaker to protect it. You could also add electrical noise filters and a neon lamp wired to the GFCI receptacle.

I think the GFCI receptacle is essential even though all of my old amps have only two-wire plugs. It is a good idea to replace those old unpolarized two-wire cords with modern two-wire polarized cords. If necessary, rewire the amp's switch, fuse, and transformer so that there are no breaks in the neutral conductor. I like to put the fuse in the circuit first, ahead of the switch. ❖

Michael Sciascia
Livingston, N.J.

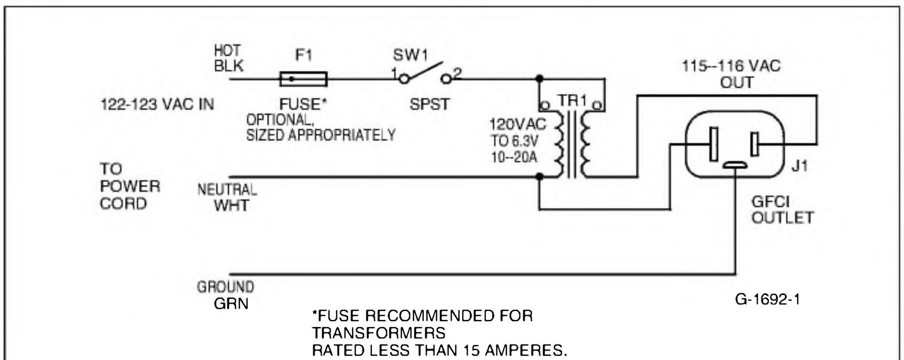

FIGURE 1: Line voltage conditioner.

Test Tracks

Featuring reader-submitted music selections used to test audio systems.

1. Taj Mahal, *Recycling the Blues and Other Related Stuff*, Columbia 31605 (LP), "Texas Woman Blues," "Sweet Home Chicago," "Cakewalk in to Town."
This is my favorite recording for judging the natural reproduction of the human voice. The Pointer Sisters provide back-up vocals on the first two tracks, and Taj accompanies himself with only a resophonic guitar ("Sweet Home Chicago"), acoustic bass ("Texas Woman Blues"), or tuba ("Cakewalk in to Town"). The sparse arrangements allow you to easily hear the quality of the voice or instrument being reproduced. The acoustic bass is great for judging bass response and speed.

2. Cassandra Wilson, *Blue Light 'Til Dawn*, Blue Note CDP 0 777 7 81357 22 (CD), "Come on in my Kitchen."
Today's best jazz vocalist? I love the unique quality of her voice and the very unusual instrumentation on this breakthrough CD. In addition to Cassandra's voice and hand claps, I listen for the reproduction of the acoustic bass, transients of the plucked mandolin, and impact of the drum tracks on this fine Robert Johnson cover.

3. Leon Parker, *Belief*, Columbia CK 67457 (CD), "Close Your Eyes."
Another spare instrumentation (acoustic bass, alto sax, and drums) allows you to clearly hear how natural the reproduction is. Can you identify when Leon switches to playing with his hands during his short drum solo?

4. Ricki Lee Jones, *Pirates*, Warner Bros. BSK-3432 (LP), "We Belong Together."
Dynamics! The soft to loud range on this track is a challenge for hi-fi audio systems. The explosive drum break and cymbal crash are some of the most realistically recorded percussion I've heard.

5. Hampton Hawes Quartet, *All Night Sessions, Volume 1*, Contemporary OJCD-638-2 (S-7545) (CD), "Jordu."
This is the first of three volumes from a legendary 'til daybreak LA recording session with Jim Hall (guitar), Red Mitchell (bass), and Bruz Freeman (drums) in addition to Hawes on piano. It is very difficult for an audio system to get piano right with the percussive attack and decay, and complex harmonic overtones. This is one of my favorite piano recordings. I also like the quality of the stereo on this recording, which allows the instruments space within a defined room.

6. Aaron Neville, *Warm Your Heart*, A&M 75021 5354 2 (CD), "It Feels Like Rain," "Louisiana 1927," "Everybody Plays the Fool."
One of rock's most unique and beloved voices sings great songs by John Hiatt, Randy Newman, and others. In addition to the quality of Aaron's falsetto, I like to listen for the deep bass and Ry Cooder's slide guitar ("It Feels Like Rain"), massed vocal choir ("Louisiana 1927"), and the variety of percussion ("Everybody Plays the Fool").

7. Take Six, *Take Six*, Reprise 0-25670-1 (LP), "Goldmine," "Get Away Jordon," "Mary."
Who doesn't enjoy the virtuosity and verve of this gospel/jazz a cappella sextet? Even though the recording is somewhat "processed," the vocal qualities of each singer still come through. I listen for how well a system can keep the six vocal lines distinct and deliver the rhythmic drive of these tracks. ❖

Paul Spiegal
South Pasadena, Calif.

Let's hear from you. Simply describe your seven favorite pieces (not to exceed 1000 words); include the names of the music, composer, manufacturer, and manufacturer's number; and send to "Test Tracks," Audio Amateur, Inc., Box 876, Peterborough, NH 03458. We will pay a modest stipend to readers whose submissions are chosen for publication.