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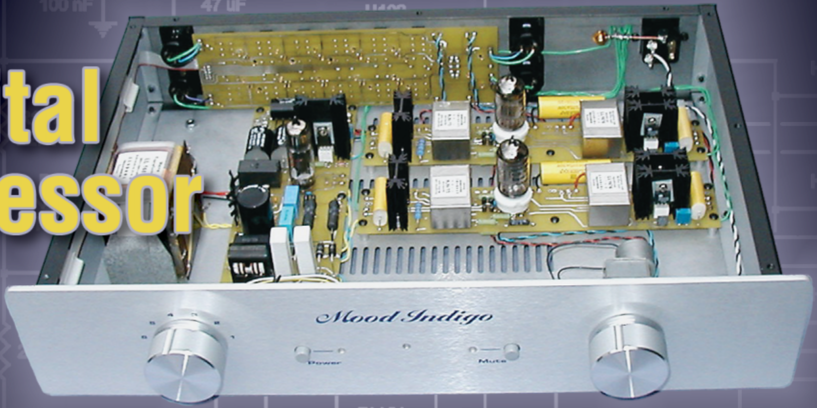

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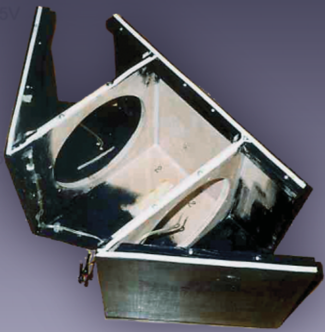

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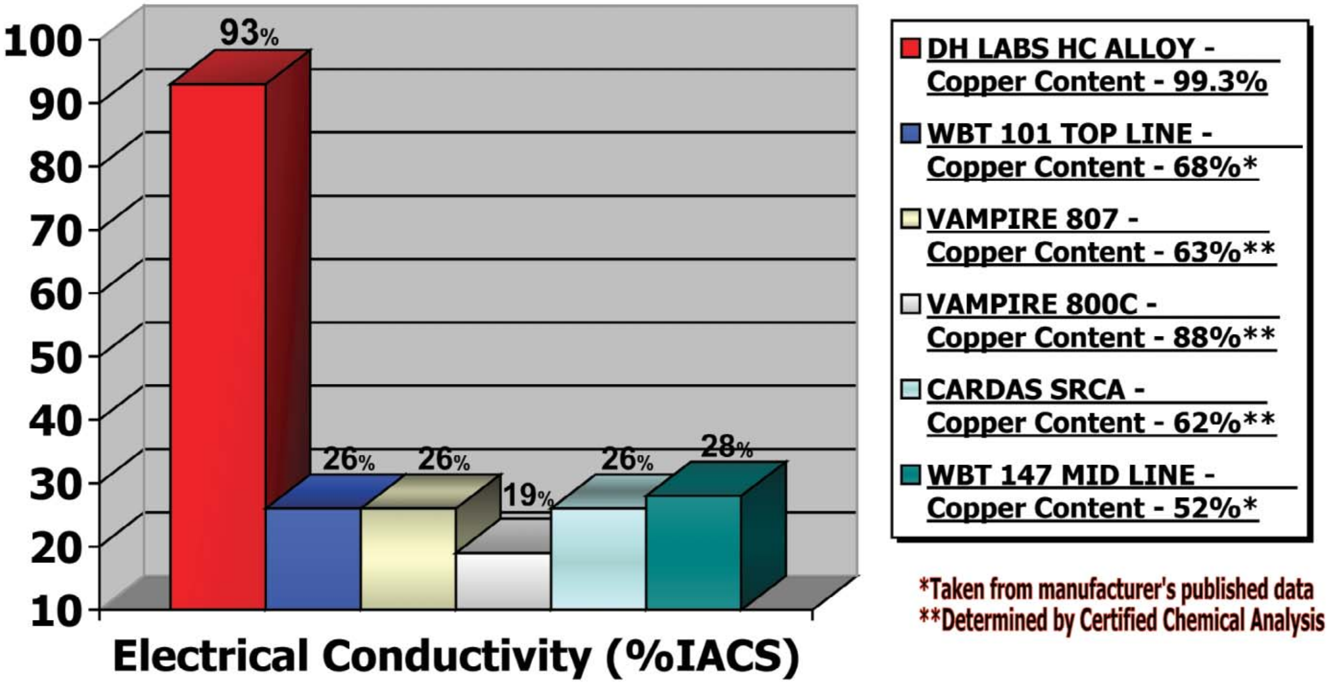

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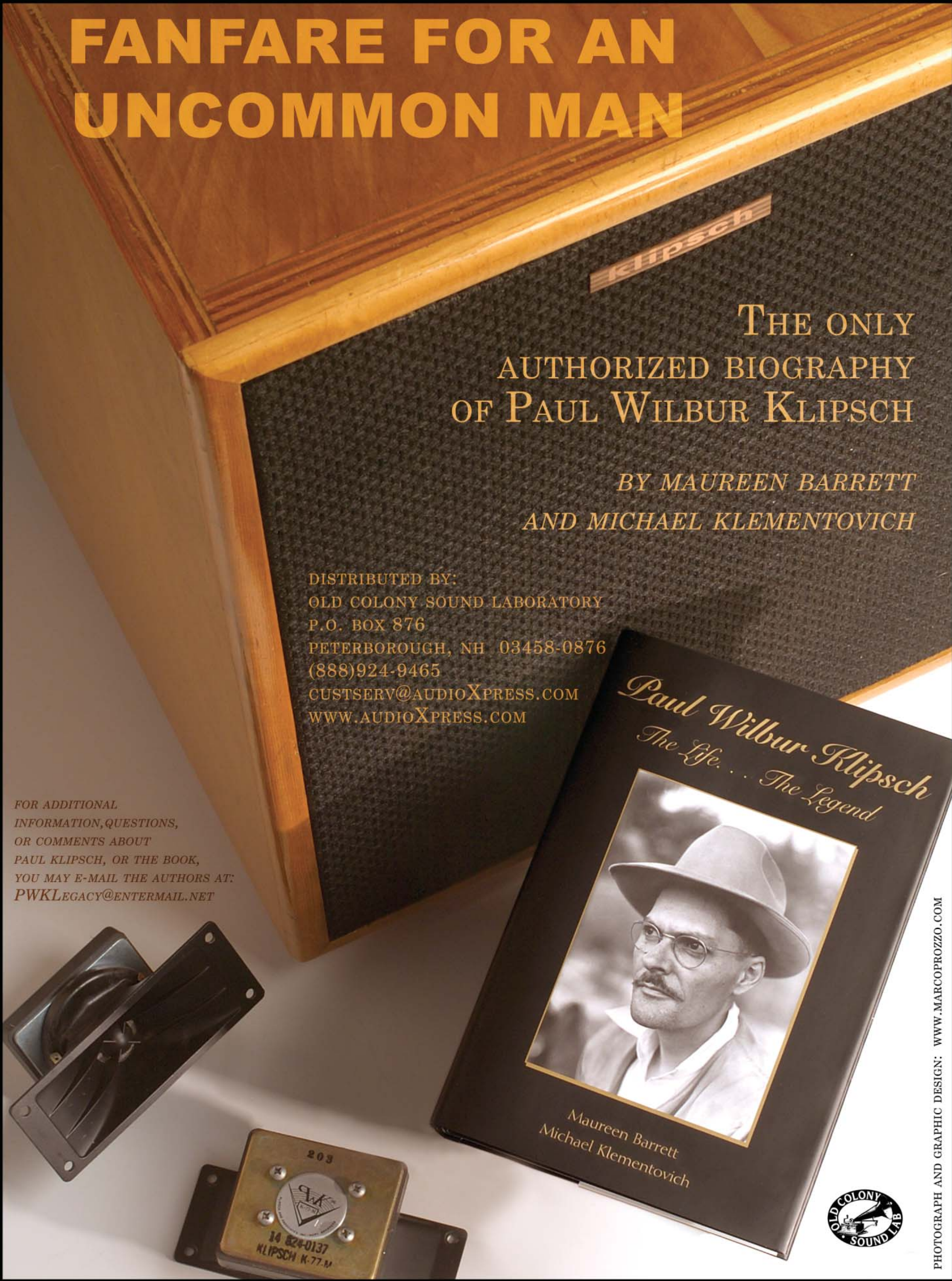
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The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it— **John Stuart Mill**

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The MaxxBass Subwoofer

With this dedicated processor, you can experience low bass notes from a relatively small speaker.

By Ron Tipton

PHOTO 1: Model 907 front panel. Signal in and out connectors and DC power in connector are on the rear panel.

MaxxBass[1] (which I'll abbreviate as MB in this article) is a *psychoacoustical* method of producing sub bass. That is, the low-frequency notes you hear don't exist except in your perception, so you can use a small driver in a small enclosure. This phenomenon has been known for hundreds of years, so I'll first cover a little history.

During the Middle Ages, many large organs with 32¢ pipes were installed in European churches. These instruments could produce a truly awesome amount of bass, and composers wrote music to take advantage of this. But problems arose when pieces began to fall off the buildings from the high intensity sonic energy. So, many of the long pipes were "stopped" so they could not be played. But people, then as now, loved that awesome sound.

By the early 1700s, some composers discovered they could trick the listener into hearing bass notes that weren't really there by playing a combination of higher frequencies. Leo Beranek apparently coined the phrase, the "Phenomenon of the Missing Fundamental," when he wrote about this in his 1993 book *Acoustics*[2]. His investigations showed that the perceived pitch of a combination of tones equally spaced in frequency is the constant difference frequency. This constant difference is the missing fundamental.

Waves, Inc. has patented an algorithm, which they call MaxxBass, to implement this technique. They have incorporated the algorithm in an easy-to-use dedicated processor which also includes an input analog-to-digital con-

PHOTO 2: MaxxBass test system showing the peak-to-peak voltmeter, dual tunable filter, and the Model 907.

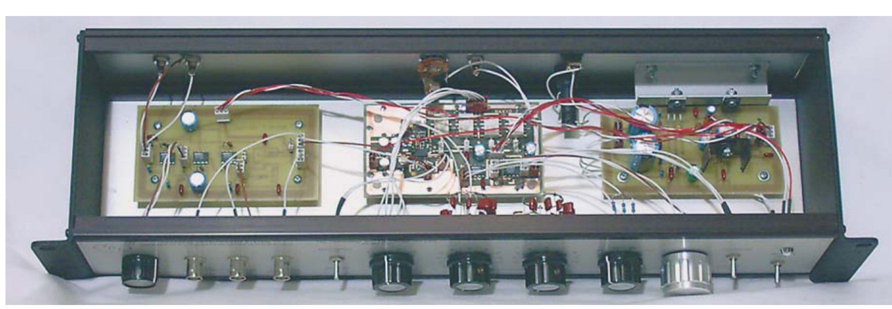

PHOTO 3: Model 907 internal view. The power op amp and +5V regulator on the right-side power supply board are heatsinked to the rear panel with a 3.5†length of ¾ · ½†aluminum channel.

ABOUT THE AUTHOR

Ron Tipton has degrees in electrical engineering from New Mexico State University and is retired from an engineering position at White Sands Missile Range. In 1957 he started Testronic Development Laboratory (now TDL Technology, Inc., 5260 Cochise Trail, Las Cruces, NM 88012, 505-382-3173, www.zianet.com/tdl) to develop audio electronics. During the 1960s and 70s, TDL built active filters and pseudo-random noise generators for well-known companies such as Bose Corp. and Acoustic Research. He is still the TDL president and principal designer.


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verter (ADC), an output digital-to-analog converter (DAC), and an output low-pass filter whose cutoff frequency you can set with an external capacitor. This processor, the MX3000AS, is priced at about ten dollars in small quantities[3] and is the "heart" of the MB development system I designed and built to "play" with this method.

THE HARDWARE

I called this unit Model 907 (*Photo 1*). The block diagram (*Fig. 1*) shows the signal flow.

The input amplifier combines the two stereo channels so the signal is single-channel (mono) thereafter. A control to set the input level is needed because the MX3000AS processor is limited to a maximum input of 2V peak-to-peak (p-p). (The data sheet says 2V p-p but in e-mail correspondence the manufacturer recommends 1V p-p for best performance.) A front-panel connector for a p-p voltmeter is included to help set the input level. Depending on the settings of the other controls, I can hear a noticeable amount of bass enhancement at input levels of 100mV p-p and above.

Front-panel connectors are provided to use an external low-pass filter because it's useful to be able to vary the cutoff frequency. Usually the low-pass cutoff frequency is set to match your main speaker's low-frequency cutoff, but it can be instructive to vary this and listen to the result. The circuit board does, however, include space for an internal low-pass filter.

A dual op amp and associated components will implement a 4-pole Sallen-Key (voltage-follower) low-pass Butterworth filter with your choice of cutoff frequency[4]. (I left this portion of the board unpopulated in my prototype unit.) A pair of easy on-off jumpers on the circuit board selects either the internal or external filter. *Photo 2* shows my test system.

A front-panel toggle switch turns the MB processor on or off. There is still a signal path through the processor when it's off so you can easily hear the difference. Also, you should turn off the processor for a few seconds whenever the clock frequency is changed, because this switch is also the reset input. I'll cover how to choose the clock frequency later.

The intensity controls (two 11-position rotary switches) set the cutoff frequency of the MB processor's internal low-pass filter by changing the value of a capacitor. I'll also cover how to set these switches later in the testing section. The proces-

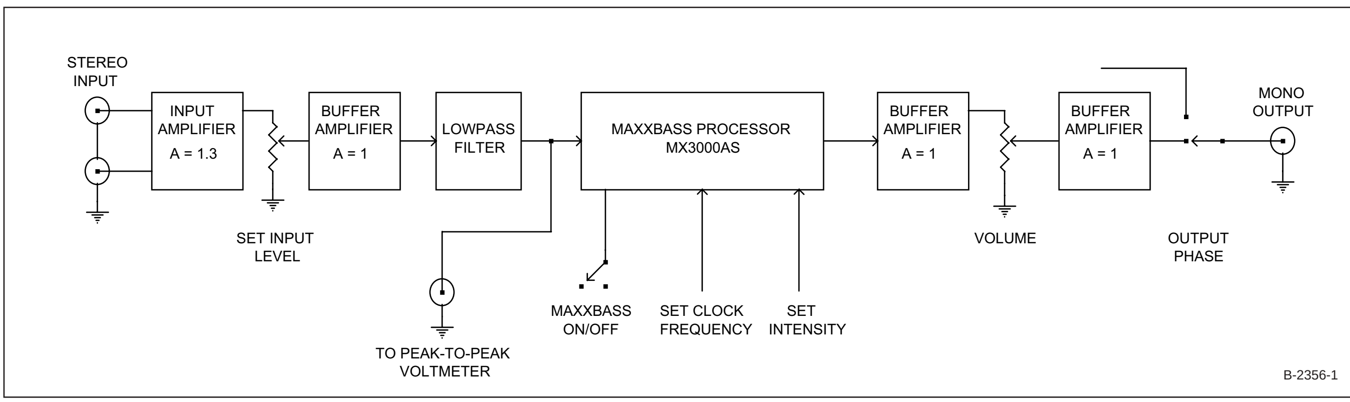

FIGURE 1: Simplified Model 907 block diagram. The low-pass filter can be either internal or external. Selection is made by jumpers on the circuit board.

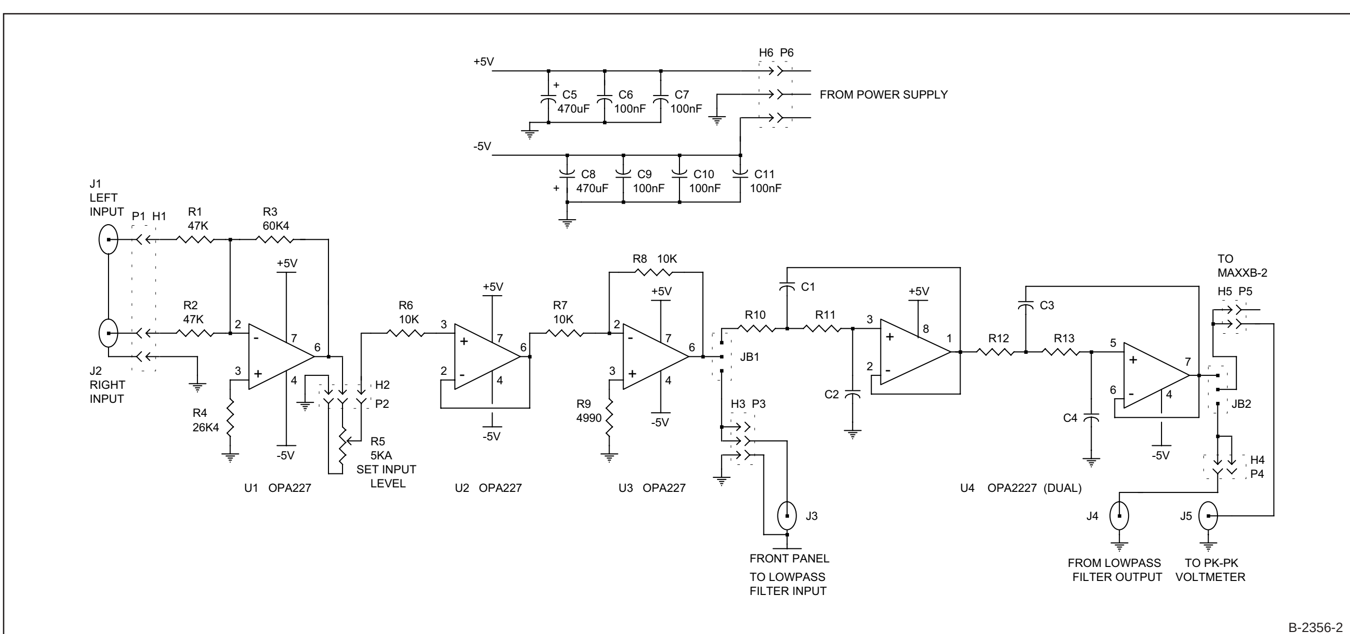

FIGURE 2: Model 907 input amplifier and optional internal low-pass filter. JB1 and JB2 select either the internal or external filter.

sor is followed by a unity gain buffer amplifier, volume control, and output buffer amplifier. The signal path through the Model 907 is non-inverting when the internal low-pass filter is used. However, if you use an external low-pass filter, it might be inverting, so I included a phase-reversing toggle switch at the output.

The circuit diagrams are included as *Figs. 2-6* and there is a complete parts list as well. *Photo 3* is an internal view showing how the circuit boards are positioned. The single-sided board on the left is the input amplifier and unpopulated space for the internal low-pass filter. The center-board contains the MB processor and output buffers. It is a double-sided board which is mostly ground plane on its bottom side. I built a single-sided processor board, but it oscillated so it appears that a double-

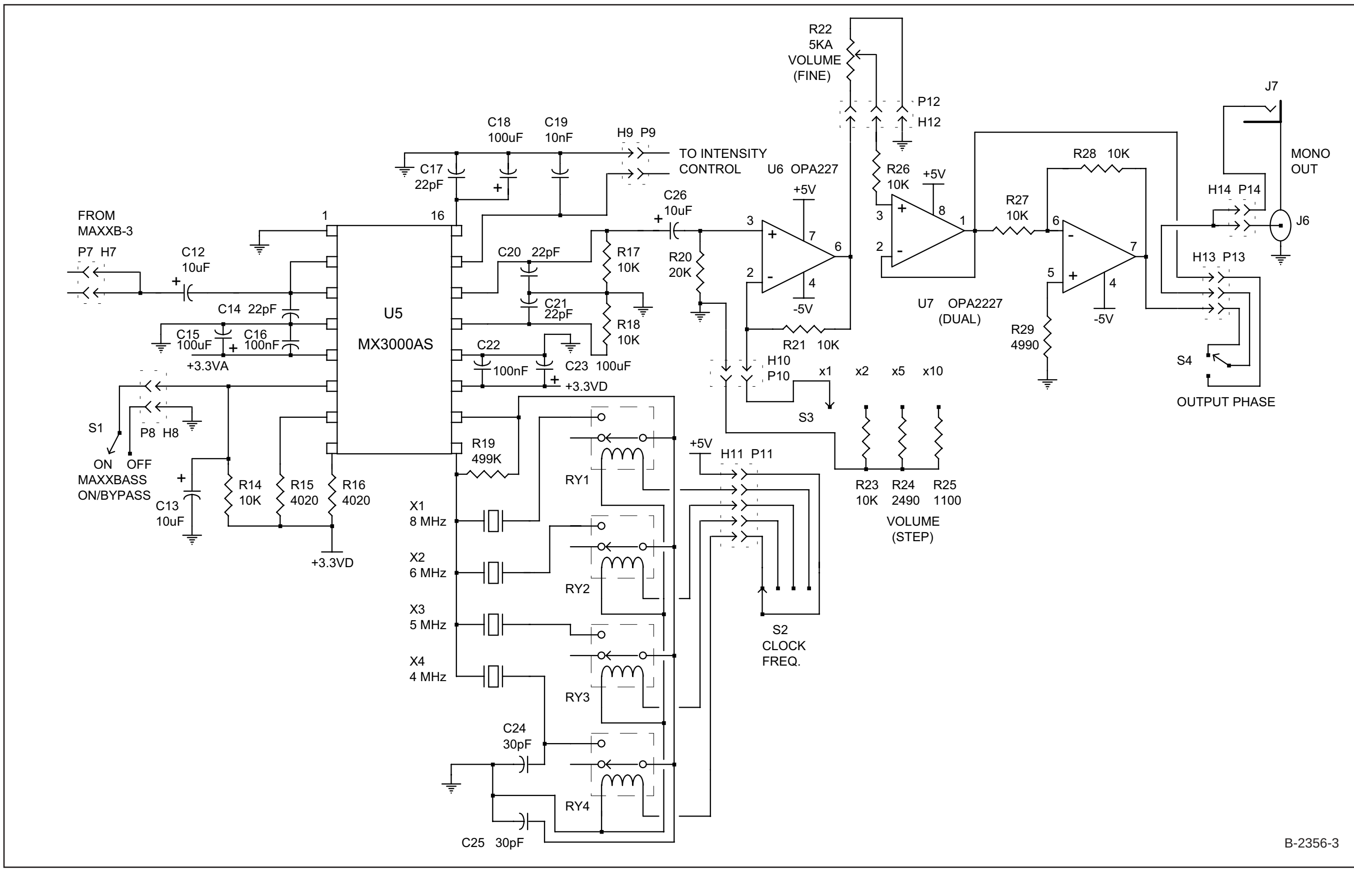

FIGURE 3: Model 907 MaxxBass processor and output buffer amplifier. All components around U5, including the resonators and relays, are surface mount.

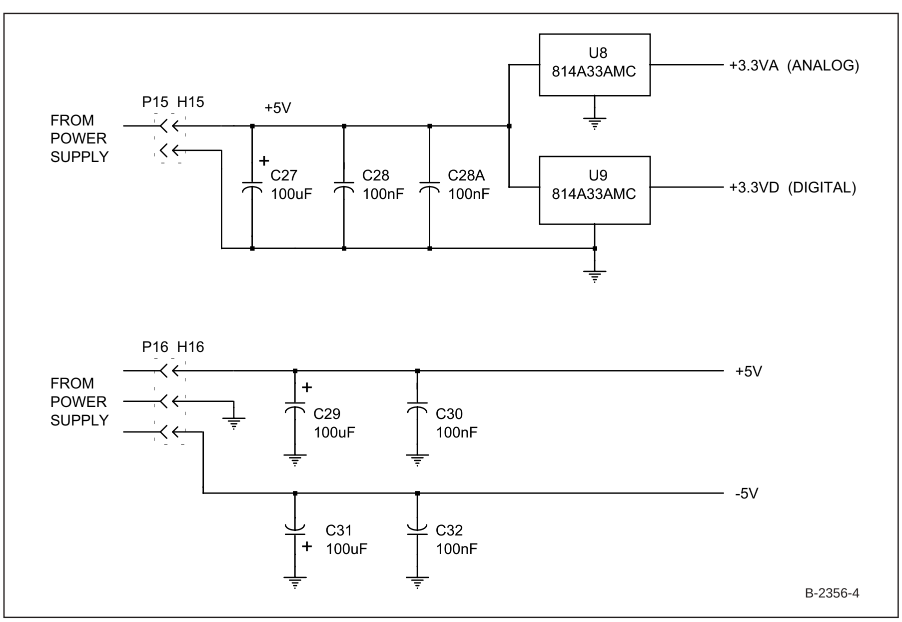

FIGURE 4: These parts are on the MAXXB-2 circuit board. Dual +3.3V regulators provide good isolation between the analog and digital signals.

sided board is needed.

Note that all the components associated with the surface-mount processor are also surface mount. The right-hand board is the power supply that provides regulated –5V to the other boards from a 24V DC wall supply.

CONSTRUCTION NOTES

The front and rear enclosure panels are anodized aluminum, which is a non-conductive surface. Connector lock washers may or may not bite through, so it's useful to use a high-speed tool (Dremel or equivalent) with a ⅛† diameter burr to remove the anodize from the area under the lock washers.

The rear panel is connected to circuit ground with a single wire from an input connector ground lug to the center pin of P1 on the input circuit board, MAXXB-3. The front panel is connected to ground with a single wire from a front-panel ground lug to P3. A single connection from each panel to the input circuit board helps avoid ground loops. (The prototype unit is very quiet, as you can see from the spectrum in *Fig. 8A*.)

The intensity control capacitors (*Fig. 5*) are mount-

PARTS LIST

MODEL 907 MAXXBASS DEVELOPMENT SYSTEM

REFERENCE	VALUE	DESCRIPTION	MANUFACTURER
R1, R2	47k	1%, ¼W, metal film	
R3	60k4	1%, ¼W, metal film	
R4	26k4	1%, ¼W, metal film	
R5, R22	5k	Audio taper potentiometer	Mouser31VJ305
R6, R7, R8, R21,R23, R26, R27, R28	10k	1%, ¼W, metal film	
R9, R29	4990	1%, ¼W, metal film	
R10, R11, R12, R13	Value depends on cutoff frequency of the internal low-pass filter (see text)		
R14, R17, R18	10k	1%, ¼W, surface mount, 1206	
R15, R16	4020	1%, ¼W, surface mount, 1206	
R19	499k	1%, ¼W, surface mount, 1206	
R20	20k	1%, ¼W, metal film	
R24	2490	1%, ¼W, metal film	
R25	1100	1%, ¼W, metal film	
R101, R102	22k1	1%, ¼W, metal film	
R103	1000	1%, ¼W, metal film	
R104	15k	1%, ¼W, metal film	
R105	1W	5%, 1W, carbon film	
R106	374	1%, ¼W, metal film	
C1, C2, C3, C4 (see text)	Value depends on cutoff frequency of the internal low-pass filter		
C5, C8, C109, C111	470m F	25V, radial electrolytic	
C6, C7, C9, C10 C30, C32, C42, C102, C104, C106, C108, C110	100nF	50V, 5%, polyester film	
C11	not used		
C12, C13, C26	10m F	10V, surface mount tantalum, 1210	
C14, C17, C20, C21	22pF	50V, 5%, surface mount COG ceramic, 1206	
C15, C18, C23, C27	100m F	10V, surface mount electrolytic	
C16, C22, C28, C28A	100nF	50V, 5%, surface mount ceramic, 1206	
C19	10nF	50V, 5%, surface mount COG ceramic, 1206	
C24, C25	30pF	50V, 5%, surface mount COG ceramic, 1206	
C29, C31, C107	100m F	25V, radial electrolytic	
C33	560nF	50V, 5%, polyester film	
C34	470nF	50V, 5%, polyester film	
C35	390nF	50V, 5%, polyester film	
C36	330nF	50V, 5%, polyester film	
C37	270nF	50V, 5%, polyester film	
C38, C112	220nF	50V, 5%, polyester film	
C39	180nF	50V, 5%, polyester film	
C40	150nF	50V, 5%, polyester film	
C41	120nF	50V, 5%, polyester film	
C43, C53	10nF	50V, 5%, polyester film	
C44	68nF	50V, 5%, polyester film	
C45	56nF	50V, 5%, polyester film	
C46	47nF	50V, 5%, polyester film	
C47	39nF	50V, 5%, polyester film	
C48	33nF	50V, 5%, polyester film	
C49	27nF	50V, 5%, polyester film	
C50	22nF	50V, 5%, polyester film	
C51	18nF	50V, 5%, polyester film	
C52	15nF	50V, 5%, polyester film	
C101, C102	2200m F	35V, radial electrolytic	
C105	47m F	25V, radial electrolytic	
U1, U2, U3, U6	OPA227P	Single op amp, 8-pin DIP	
U4, U7	OPA2227P	Dual op amp, 8-pin DIP	
U5		MX3000AS 16-pin SOIC	MaxxBass processor, Waves, Inc.
U8, U9	814A33AMC	+3.3Vregulator,5-pin SOT-23-5	

REFERENCE	VALUE	DESCRIPTION	MANUFACTURER
U101		LM675T Power op amp, TO-220	
U102	7805A	+5V regulator, TO-220	
U103	7905A	-5V regulator, TO-220	
D101		Red LED Power-on indicator	Lumax SSI-LXR1612ID (Digi-Key 67-1147)
J1, J6		Female, panel mount RCA, black	DGS (Mouser 161-1052)
J2		Female, panel mount RCA, red	DGS (Mouser 161-1053)
J3, J4, J5		Female, panel mount BNC	AMP (Mouser 523-221-RFX)
J7		Mono phone jack, ¼†	Switchcraft (Mouser 502-11)
J101		2.5mm male, insulated, panel mount, power input connector	DGS (Mouser 163-4303)
P1, P2, P3, P6, P12, P13, P16, P103, P104, P105		3-pin shell with terminal pins	Molex WM2012
P4, P5, P7, P8, P9, P10, P14, P15, P101, P102, P106		2-pin shell with terminal pins	Molex WM2011
P11		5-pin shell with terminal pins	Molex WM2014
		Terminal pins for the Molex shells	Molex WM2200
H1, H2, H3, H6, H12, H13, H16, H103, H104, H105		3-pin male header	Molex WM4001
H4, H5, H7, H8, H9, H10, H14, H15, H101, H102, H106		2-pin male header	Molex WM4000
H11		5-pin male header	Molex WM4003
JB1, JB2		Jumper block (3-pin male header) with 2 jumpers	Molex WM4001
S1, S4, S101		SPDT miniature toggle switch	
S2		Rotary switch, 1-pole, 4-positions, break-before-make	Mouser 105-14571
S3		Rotary switch, 1-pole, 4-positions, make-before-break	Mouser 105-13571
S5, S6		Rotary switch, 1-pole, 11-positions, make-before-break	Mouser 105-13571
RY1, RY2, RY3, RY4		DPDT relay, 5V coil, 190W , surface mount	NEC EB2-5NU
X1		Ceramic resonator, 8MHz, surface mount	Panasonic (DigiKey PX800PCT)
X2		Ceramic resonator, 6MHz, surface mount	Panasonic (DigiKey PX600PCT)
X3		Ceramic resonator, 5MHz, surface mount	Panasonic (DigiKey PX500PCT)
X4		Ceramic resonator, 4MHz, surface mount	Panasonic (DigiKey PX400PCT)
		Rack mount enclosure, 3.5 ·5 ·19†	Sescom 2RU5
		Front panel	Metalphoto of Cincinnati, PN50152)
		Knob for Set Input Level	Jameco 138392CA
		Knobs (4) for rotary switches	Eagle (Mouser 45KN017)
		Knob for Volume control	Jameco 138510CA
		Circuit board MAXXB-2	
		Circuit board MAXXB-3	
		Circuit board PWRMAXX	
		Heatsink for 7905A regulator	Jameco 228005CA
		Misc. hardware, teflon insulated wire, and shrink tubing	
	24V DC @ 100mA	Wall DC power supply,	Mouser 412-124013

ed on the rotary switches S5 and S6. This is made easier by bending a ⅞† diameter circle of #20 gauge solid wire around a pill container or similar object. Then, solder the capacitors between the switch contacts and the wire circle, making it the common contact.

The double-sided circuit board, MAXXB-2, is designed for non-plated through holes so there are quite a few vias so you can solder components such as the Molex headers on the bottom side. The top and bottom side are simply connected with a short length of solid wire soldered to both sides.

The three boards are mounted on the bottom enclosure plate with 4-40 · ⅜† nylon spacers and 4-40 machine screws. You can use metal spacers for the input and power-supply boards, but non-conductive hardware is essential for the processor board to avoid ground loops.

You can download circuit board layouts and parts placement drawings from the TDL website[5]. I designed the boards using CIRCAD, and, you'll need to download a free MS-DOS version from the Holo-phase website at www.holo-phase.com to print them.

A LITTLE THEORY

Figure 7 shows input and output spectra from the MX 3000AS data sheet for a 50Hz input signal. The output has harmonics spaced 50Hz, which is the missing fundamental. *Figure 8* shows the spectra I measured using a 50Hz

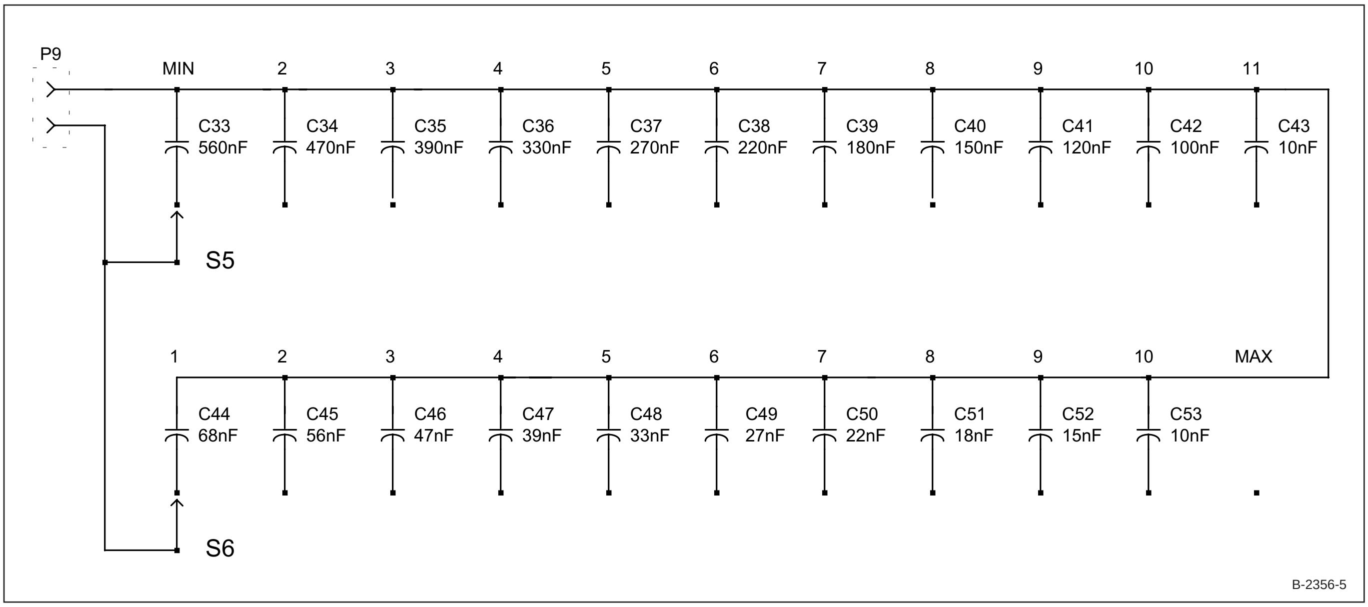

FIGURE 5: The MaxxBass intensity controls. The capacitors are mounted on the rotary switches.

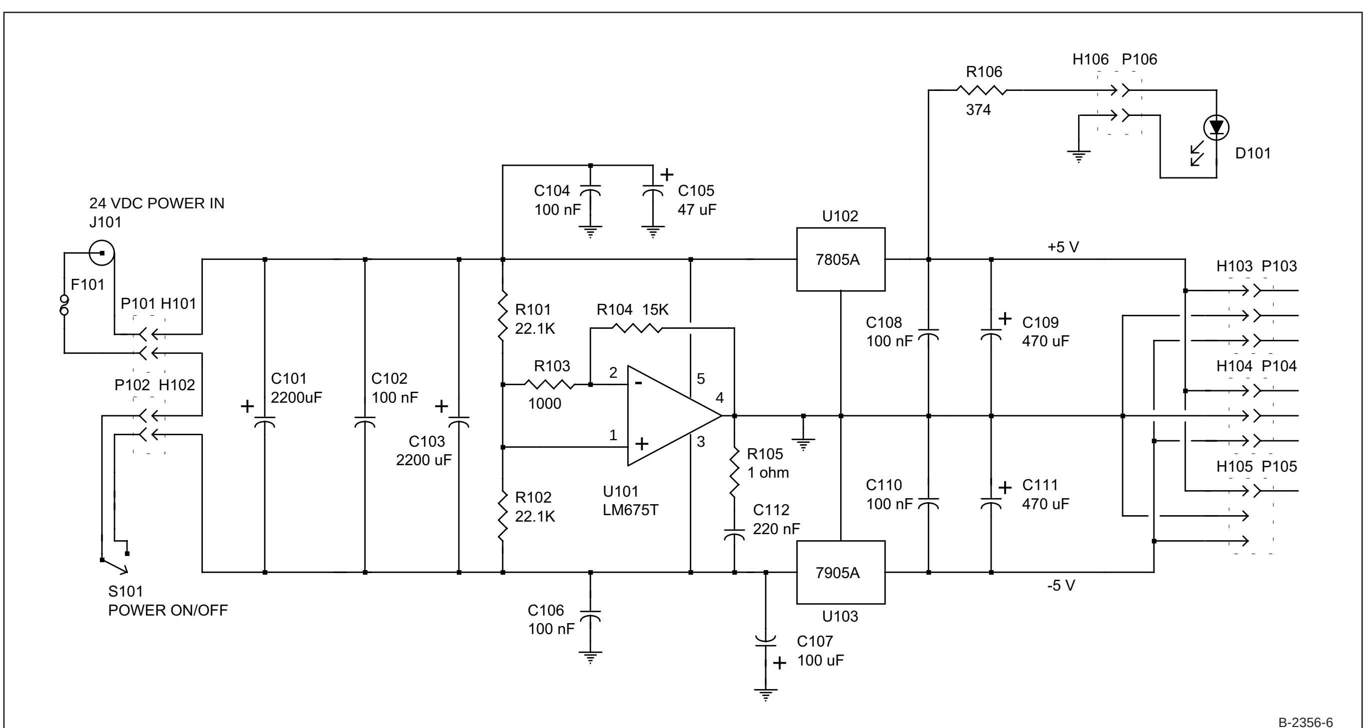

FIGURE 6: Model 907 power supply. The power op amp (U101) "splits" the 24V DC input into −12V.

input and the intensity control set for maximum for a speaker with a cutoff frequency of 85Hz. This output shows lots of harmonics at 50Hz and other spacings which indicates the data sheet output spectrum is somewhat simplified.

There are some relationships you need to look at before going on to the

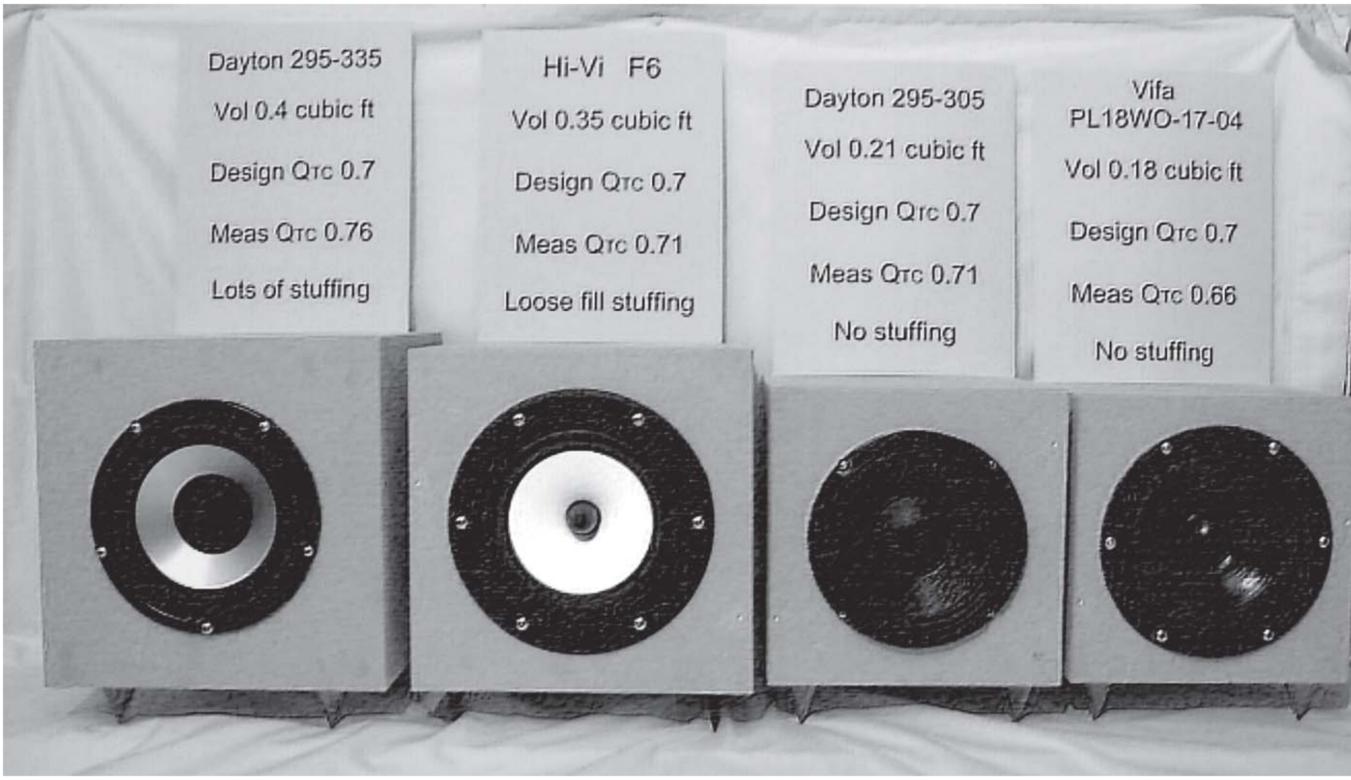

PHOTO 4: The four MaxxBass test speakers. All have a measured closed box Q close to 0.7[9].

testing section. First, the MB cutoff frequency, fMB, is set to about 80% of the speaker's cutoff frequency (fc). That is:

$$FMB = 0.8\ fc$$

(The data sheet says 70% to 90%, so 80% seems like a good choice.)

Next, for maximum intensity, the processor's output low-pass filter external capacitor value should be calculated for a cutoff frequency about ten times fMB. That boils down to:

$$CLP = (8 \cdot 10^{-6})/fMB \text{ (in microfarads)}$$

Finally, the clock frequency, (fclock), is calculated from:

$$FCLOCK = 102400 \cdot fMB$$

(The clock frequency can vary considerably from the calculated value without any noticeable change in performance.)

TESTING

Even though I understand that this technique does work, it was still startling to hear low notes coming from a small box when I first turned the system on. MB speakers have been built with 5† drivers, but I wanted to move a bit more air. *Table 1* lists a number of 7† (nominal) drivers—all of which have a Q

TABLE 1: 7†DRIVER COMPARISON FOR MAXXBASS

MFG	MODEL	POWER	FC@Q=0.7	VOLUME	SPL	PRICE
Audax	AP170ZO	60W	89Hz	0.37ft^3	89dB	$31.80
Dayton	295-305	50W	72Hz	0.21ft^3	87dB	$17.70
Dayton	295-335	50W	55Hz	0.45ft^3	85dB	$17.65
Focal	7K4412	90W	87Hz	0.41ft^3	91dB	$117.07
Focal	7W4411	90W	78Hz	0.43ft^3	89dB	$151.23
Hi-Vi	F6	45W	81Hz	0.34ft^3	88dB	$58.50
Hi-Vi	M6N	45W	72Hz	0.45ft^3	88dB	$45.68
Peerless	832732	150W	85Hz	0.24ft^3	90dB	$33.30
Peerless	850439	150W	90Hz	0.22ft^3	88dB	$56.50
Peerless	850467	150W	76Hz	0.32ft^3	87dB	$63.30
Scan-Speak	18W-8545	100W	88Hz	0.21ft^3	88dB	$142.80
Scan-Speak	18W-8545K	100W	68Hz	0.35ft^3	87.5dB	$146.90
Scan-Speak	18W-4546	100W	81Hz	0.24ft^3	88dB	$140.50
Vifa	PL18WO09-08	100W	78Hz	0.27ft^3	87dB	$63.50
Vifa	MG18WK09-08	50W	70Hz	0.49ft^3	86.5dB	$47.90
Vifa	PL18WO17-04	100W	83Hz	0.18ft^3	89dB	$24.00*

* Sale price, not currently available but I included it because I had built an enclosure for it.

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Our equipment utilizes the best testing systems - B&K, TEF, MIssa, Clio and LMS to ensure perfect quality.
We also use advanced software including Ansoft, Redrock, Protel, 3D Max and Solidworks for cutting edge design.
There are over six hundred dealers and over 200 our own retail shops within China.
Our products undergo rigorous design and innovative engineering by professionals in the US and manufactured in China.
SWANS SPEAKER SYSTEMS, INC. 2550 Corporate Place, Suite C-103, Monterey Park, CA 91754 USA
Tel: (323)-881-0606 Fax: (323)-881-0581 e-mail: sales@dulcet.com
web: www.swanspeaker.com SWANS SPEAKER a division of Hi-Vi RESEARCH®

of 0.7 in a closed box with a volume of ½ft^3 or smaller. I chose four drivers from this list and built the enclosures using ¾†MDF.

Photo 4 shows them along with their pertinent information. Two of the four have paper cones. The Dayton 295-335 has an aluminum cone, and the Hi-Vi F6 has a Kevlar cone. I included these latter two because I wanted to see whether the increased stiffness of aluminum or Kevlar made any difference in performance.

Using the equations from the theory section and the cutoff frequency from *Table 1* for the Dayton 295-335, I'll go through a MB design example.

$F_{MB} = 0.8 \cdot fc = 0.8 \cdot 55 = 44Hz$
$C_{LP} = (8 \cdot 10^{-6})/44 = 0.182m\,F = 182nF$
(for maximum intensity)
$F_{CLOCK} = 102400 \cdot 44 = 4.5MHz$

You can set the clock frequency to either 4 or 5MHz (I can't hear any difference). C_{LP} has a 10nF capacitor on the circuit board (C19) so the intensity switches are set to add about 170nF. Refer to *Fig. 4* and set the left-hand switch to 9 and the right-hand switch to 2 or 3. This adds 176nF or 167nF, respectively, and results in maximum intensity according to the processor data sheet.

I have found maximum intensity too high for some source material. For example, pipe organ music sounds "muddy." It sounds better to me when the capacitance is doubled or even tripled. As you can see from *Fig. 4*, the capacitors on the two switches are in parallel, so the total range is 10nF to 628nF, which should accommodate any speaker and any taste.

Generally speaking, all four speakers sound fine, with perhaps a slight advantage to the aluminum and Kevlar cones. This raises a problem that is fortunately easy to fix. Referring to *Figs. 7* and *8*, you can see that the processor data sheet shows the output 50Hz signal at a lower amplitude than the 100Hz component.

Figure 8 an actual measurement shows the 50Hz level 14dB *higher* than the 100Hz this difference becomes

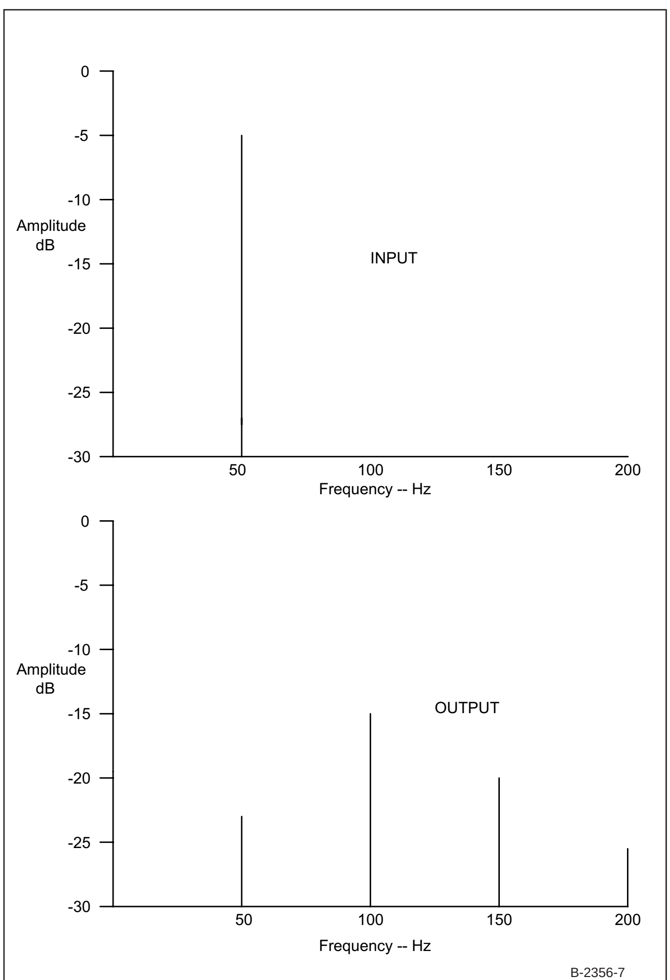


FIGURE 7: MaxxBass input and output spectra from the MX3000AS data sheet.

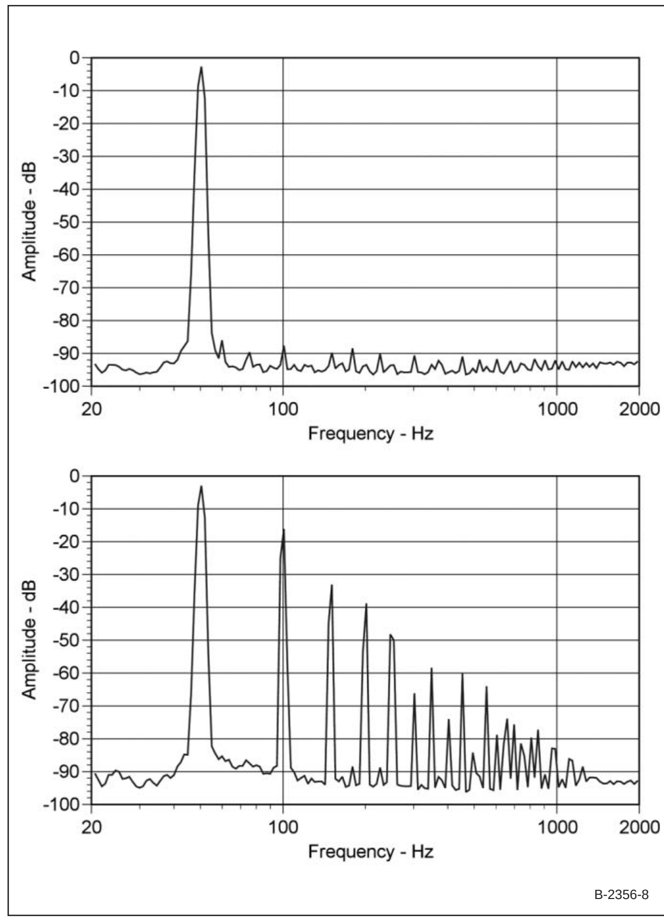


FIGURE 8: Measured input and output spectra. *TrueRTA* software was used for this measurement[8].

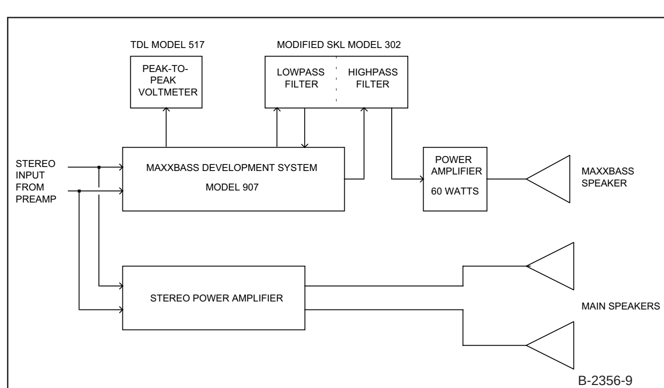


FIGURE 9: MaxxBass test setup. If used, set the high-pass filter's cutoff frequency to the MaxxBass speaker's cutoff frequency.

even larger at lower input amplitudes. This means you can drive the speaker fairly hard at a frequency below its cutoff. This can and sometimes does cause the cone to briefly break up and "buzz." The stiffer cones seem more resistant to this, but I have heard a brief buzz out of them, too.

A simple solution is to connect a tunable high-pass filter between the Model 907 output and the power amplifier input. The filter in my test setup[6] is a dual unit, so I just reconnected some cables. *Figure 9* shows the block diagram of my setup.

Sub bass is non-directional, but MB isn't sub bass in the conventional sense. The MB speaker should face forward and be located about midway between the two main speakers. Generally this works well, but it does affect stereo imaging. For example, as I was listening to a Chuck Israels Quartet CD[7], the bass (violin) seemed to come from the MB speaker in the center.

A FINAL THOUGHT

MB definitely adds a lot of "richness" to the low-frequency end. But don't be afraid to turn down the intensity until the added bass blends unobtrusively with the rest of the music. (Perhaps this is good advice for any subwoofer.) Achieving a blended intensity also varies with the type of music, so MB requires you to "ride the controls" instead of just "set it and forget it." But perhaps that's a small price to pay for a small and rather inexpensive speaker, so I'm going to keep MB in my system.

All three circuit boards will be available from FAR Circuits for $17 for the set of three boards, plus $2 postage for up to four boards. There is a $3 service charge for paying by credit card, which you can avoid by sending a check or money order. The contact information is: 18N640 Field Court, Dundee, IL 60118, phone/FAX 847-836-9148. You can download an order form in Adobe pdf format from their website at www. cl.ais.net/farcir. The double-sided board does not have plated-through holes, but neither did my prototype. Readers without computers or web connections can obtain this information by including a self-addressed #10 envelope and two US stamps.

A front panel for the Sescom rack mount case will be available from TDL Technology, Inc. for $20 postpaid in the US and Canada. The panels, which are made by Metalphoto of Cincinnati, are 0.063† thick aluminum with black lettering on a clear, anodized background. The panel is undrilled, but all hole centers are indicated by cross lines on the panel.

I'm in contact with Paul Bundschuh, the Waves semiconductor sales and marketing VP in Austin, Tex. He is very interested in this project and wants to work with me, so the MaxxBass processor will be available from TDL Technology, Inc. at cost plus $2 for packaging and postage. I don't have a price yet.

A completed and tested model 907 with power supply will be available for $283 plus $15 shipping in the US. (Please ask for a price quote for shipping to other countries.) We probably won't stock these, so delivery would be two to four weeks after receipt of an order, but this could change if demand is higher than expected. ❖

REFERENCES

1. MaxxBass is a registered trademark of Waves, Inc. The algorithm is also protected by US patent 5,930,373. You can find the MX3000AS data sheet and several technical papers and application notes on their website at www.waves.com.
2. Beranek, Leo, *Acoustics*, Acoustical Society of America, 1993.
3. Waves, Inc. has distributors in many states and in many countries. You can find a list on their website.
4. Several analog filter design programs are freely available on the Internet. For example, you can download *Filter Lab* from www.microchip.com.
5. You can download Model 907 circuit board layouts and parts placement drawings from www.zianet.com/tdl/magarts.htm.
6. The dual, tunable low-pass/high-pass filter started life as an SKL model 302. I converted it from a vacuum tube unit to an all op-amp model in 1998. An article on this conversion was published in *Nuts & Volts Magazine*, July 1999. The converted filter has a much lower hum and noise level than the original.
7. *The Bellingham Sessions, Volume 2*, The Chuck Israels Quartet. Produced by Andrew Marshall as an Audio Ideas recording, ai-cd-013. www.audioideas.com.
8. *TrueRTA* (Real Time Audio Spectrum Analyzer) is a product of True Audio, 387 Duncan Lane, Andersonville, TN 37705. www.trueaudio.com.
9. The speaker enclosures were designed using *WinSpeakerz*, another computer program from True Audio. www.trueaudio.com. Measuring closed-box Q is well covered in the literature, for example, see pages 16 through 19 in *Testing Loudspeakers* by Joseph D'Appolito, Audio Amateur Press, 1998.

The Double-Dipole Subwoofer

This author from Greece shares with us his design of an open-baffle subwoofer which works well in a corner placement.

By George Danavaras

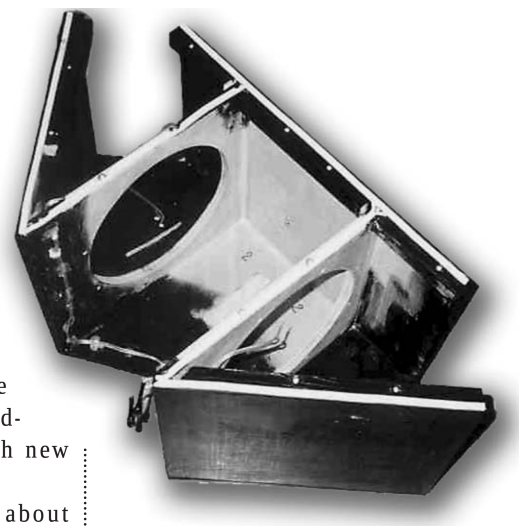


PHOTO 1: W-frame subwoofer cabinet.

Open-baffle loudspeakers offer many benefits for the reproduction of the audio spectrum. Complete freedom from internal box reflections and bipolar radiation are the most important of them. These have, as a result, openness and neutrality to the reproduction of the music that no other loudspeaker can offer.

My previous loudspeakers were full-range open-baffle speakers; each one used a 12†woofer for the bass reproduction and a large custom-made electrostatic panel for the frequencies above 600Hz. I listened to this loudspeaker for several years and was very pleased with its sound.

With the arrival of the home cinema and its multichannel sound, it was not possible to put five of these big speakers in my listening room. My first decision was to maintain two systems in my room one with the two open-baffle loudspeakers for the music reproduction and another with five small loudspeakers and a closed box subwoofer for the home cinema.

I bought five small loudspeakers and constructed a closed box subwoofer for the home cinema using two 12†woofers similar to the ones used in the open-baffle speakers. After living for some time with these two separate systems, and with the arrival of the multichannel SACD and DVD-Audio, it was obvious that the only system that could remain was the multichannel one. So I decided (one more time) to change both the loudspeakers and the subwoofer with new ones.

All that time I was thinking about building an open-baffle subwoofer since, with the closed box subwoofer I used for the home cinema system, I missed the clarity and the neutrality of the low range that I had using the open-baffle loudspeakers.

It was about that time I found the S. Linkwitz site (www.linkwitzlab.com), which contains a very good analysis of open-baffle speakers and also presents the construction of an open-baffle subwoofer and design theory for its equalization. The shape of this subwoofer was not suitable for installation in the corner of my listening room, so I designed one that was. Maybe you can install one in your room.

REQUIREMENTS

My back projection TV is in the corner of my listening room, with much space behind the box. Normally nothing can be put there, so I thought it was an ideal place for the subwoofer. First, because the corner can boost the low frequencies, and second, because it can hide the huge volume of the subwoofer.

I had four INFINITY KCS-120IB 12† woofers available from previous projects. These woofers are of excellent quality and are designed for open-baffle systems. According to the manufacturer, their features include: IMG injection-molded graphite cone, high temperature Kapton/Nomex voice coil formers, Kapton-laminated copper ribbon voice coil lead wire, and extremely high power capability.

ABOUT THE AUTHOR

George Danavaras graduated from National Technical University of Athens, Greece, in 1986 with a degree in Electronic Engineering. He currently works in the R & D division for a Greek telecommunication company. His hobbies include design and manufacturing of audio crossovers, amplifiers, and loudspeakers.

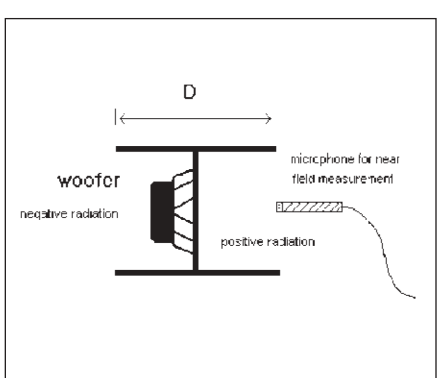


FIGURE 1: The H-frame.

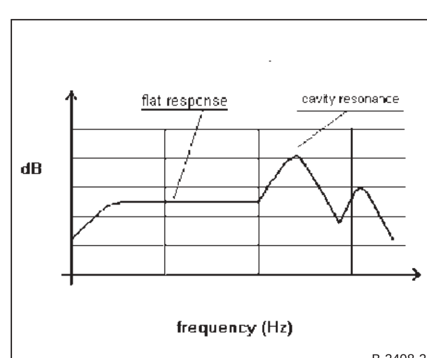


FIGURE 2: Near-field frequency response of the H-frame.

The Thiele/Small parameters of the woofers I measured were:

PARAMETER	WOOFER #1	WOOFER #2	WOOFER #3	WOOFER #4
Q_{ES}	0.735	0.712	0.808	0.72
Q_{MS}	11.23	10.14	11.72	10.23
Q_{TS}	0.69	0.66	0.75	0.67
Fs (Hz)	34	31.5	32.7	30.7
Rdc (Ohm)	3.79	3.85	3.94	4.03
Sd (square meters)	0.053	0.053	0.053	0.053
Xpeak (mm)	6.75	6.75	6.75	6.75
(According to manufacturer)				

THE OPEN-BAFFLE THEORY

One of the most important factors in the design of the open-baffle subwoofer is the distance that separates the positive and the negative sides of the woofer radiation. There are

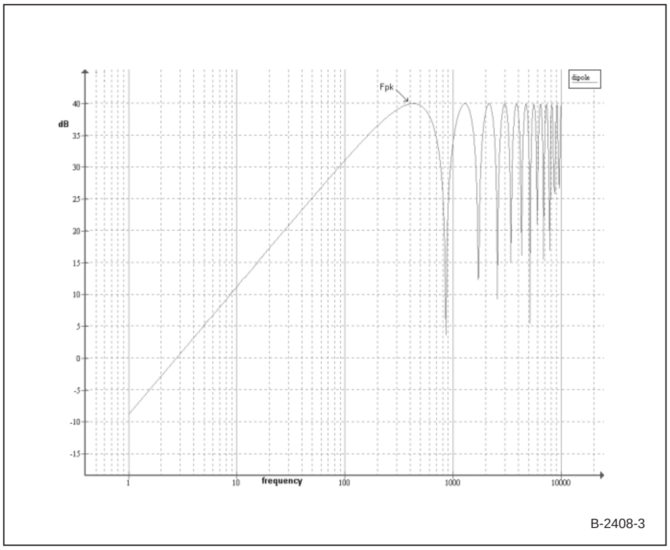


FIGURE 3: Far-field frequency response of the H-frame.

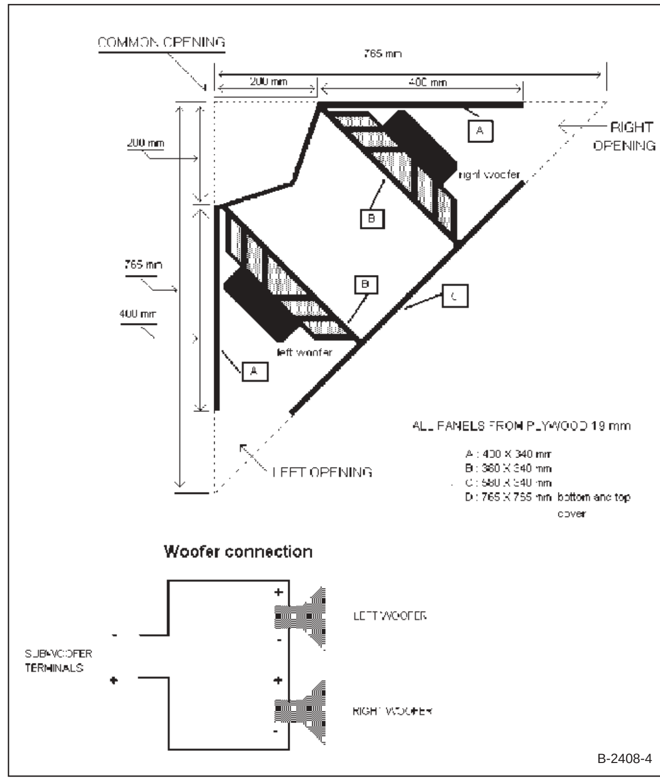


FIGURE 4: W-frame double dipole woofer.

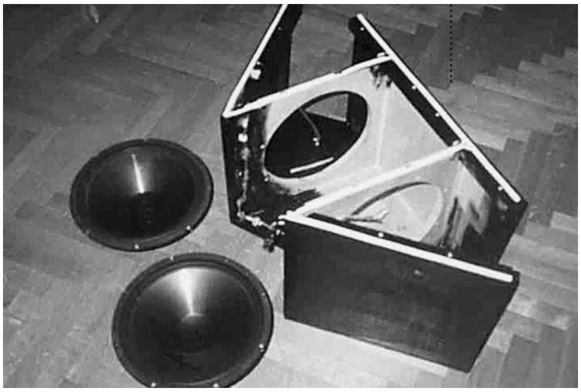


PHOTO 2: W-frame subwoofer cabinet with the two woofers.

PHOTO 3: Subwoofer installed on the corner of the listening room with the equalizer and the power amplifier below.

several different shapes that can be used as open-baffle subwoofers. The purpose of all of them is to increase the distance between the opposite sides of woofer radiation in such a way that the total size of the subwoofer remains as compact as possible.

One of the simplest shapes to perform this task is the H-frame (*Fig. 1*). The woofer is on a baffle that has extended its sides both in the front and in the back. The result is that the distance between the positive and negative sides of the woofer radiation becomes larger while the total size of the subwoofer is not much bigger.

The near-field response (with the microphone placed in front of the one opening as indicated in *Fig. 1*) is shown in *Fig. 2*. The peak in the response is caused by the 1/4 resonance of the cavity that is produced in front of (and similar behind) the woofer. In the near-field measurement, the negative side radiation of the woofer does not come into place and so the frequency response below the cavity resonance frequency remains flat down to the woofer resonance frequency.

The far-field response is shown in *Fig. 3*. The frequency Fpk where the first peak occurs is given by:

$$F_{pk} = (0.5 \cdot c)/D$$

where c is the sound velocity and D is the distance between the openings.

This peak in frequency response results when the negative polarity source after the 180 of phase shift due to the distance D adds to the positive source; the resulting pressure is twice that of a single source (+6dB). Above the peak frequency there are also many other peaks and notches for the same reason.

Below the peak frequency Fpk, the sound pressure falls at 6dB/octave due to the cancellation of the sound that comes from the negative side. At the low frequencies the distance D is small compared to the length of the sound so it cannot provide any isolation between the positive and negative sources.

The frequency at which the open-baffle

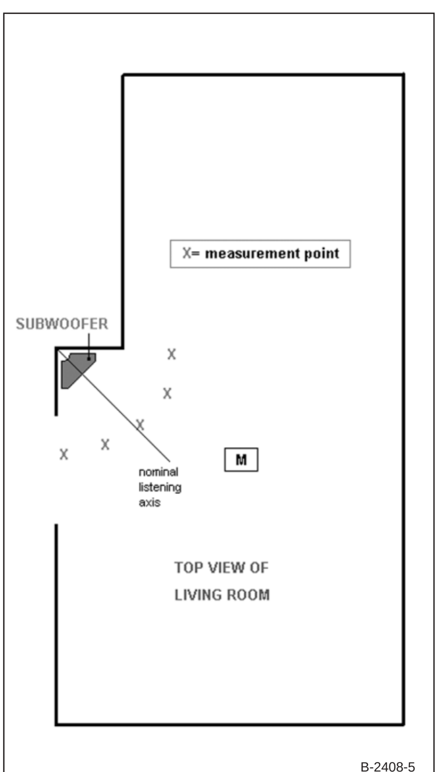


FIGURE 5: Double dipole directivity measurement.

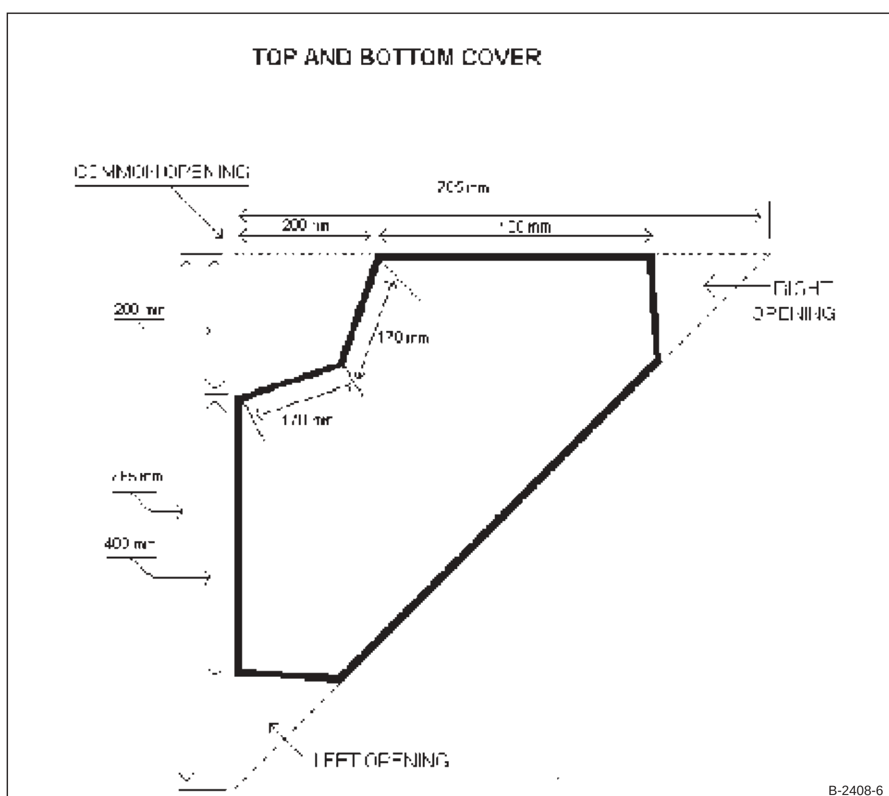


FIGURE 6: Cutout diagram for the panels D.

speaker has the same level output with a single similar monopole (for example, a closed box subwoofer) is given by:

$$F_{eq} = (0.17 \cdot c)/D$$

Above this frequency the open-baffle speaker has more output than a single monopole and below this frequency it has less output that falls at 6dB/octave where the single monopole has a flat response.

In order for the open-baffle subwoofer to be useful for music reproduction, it is absolutely necessary to equalize first the peak in the frequency response that it is due to the cavity resonance using a notch filter, and second, the far-field low-frequency rolloff. The equalizer should boost the low frequencies with 6dB/octave in order to flatten the frequency response. This boost means that the excursion of the woofer will be increased so the linearity will be worse. That is why you need a lot of drivers in order to get high volumes of sound at the low frequency with an open-baffle enclosure.

The directivity of the open-baffle subwoofer is maintained also at the lowest

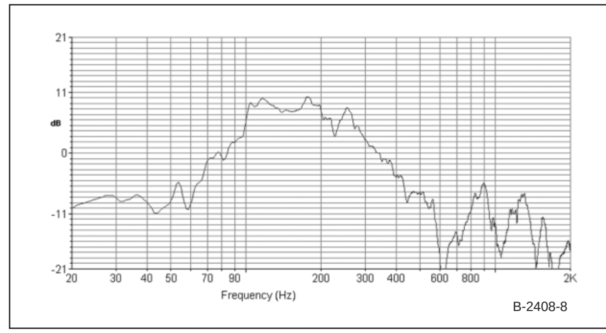


FIGURE 8: Measurement of the dipole subwoofer in the corner of the room and on the floor.

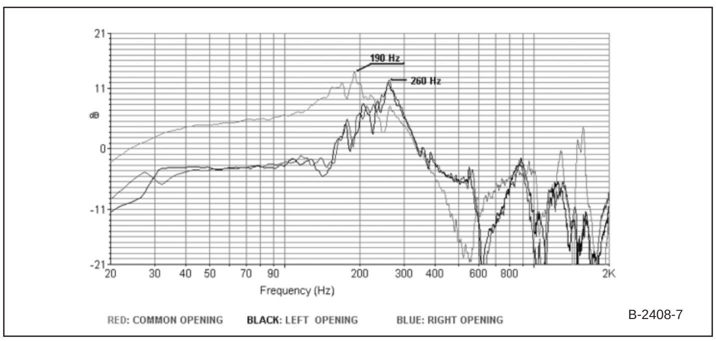


FIGURE 7: Measurement of the dipole subwoofer in the middle of the room.

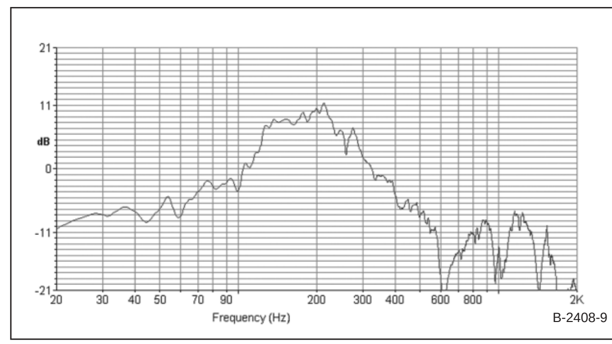


FIGURE 9: Measurement of the subwoofer in the corner of the room and 55cm above the floor. This is the final position.

frequencies. The response at all frequencies drops -3dB when the listening point is at an angle of 45 with the dipole axis and -6dB when it is at 60 . This directivity means that if you want to have the maximum sound pressure from the open-baffle subwoofer, you should remain in front of its axis.

THE NOVEL VARIATION OF THE W-FRAME

I already mentioned that I intended to install the subwoofer in the corner of my living room. Unfortunately, for this kind of installation the H-frame is not suitable since it requires much free space both to the front and to the back of the cabinet. After several thoughts and tries about which shape would be suitable for installation in the corner, I decided on the cabinet shown (without the cover) in *Photo*

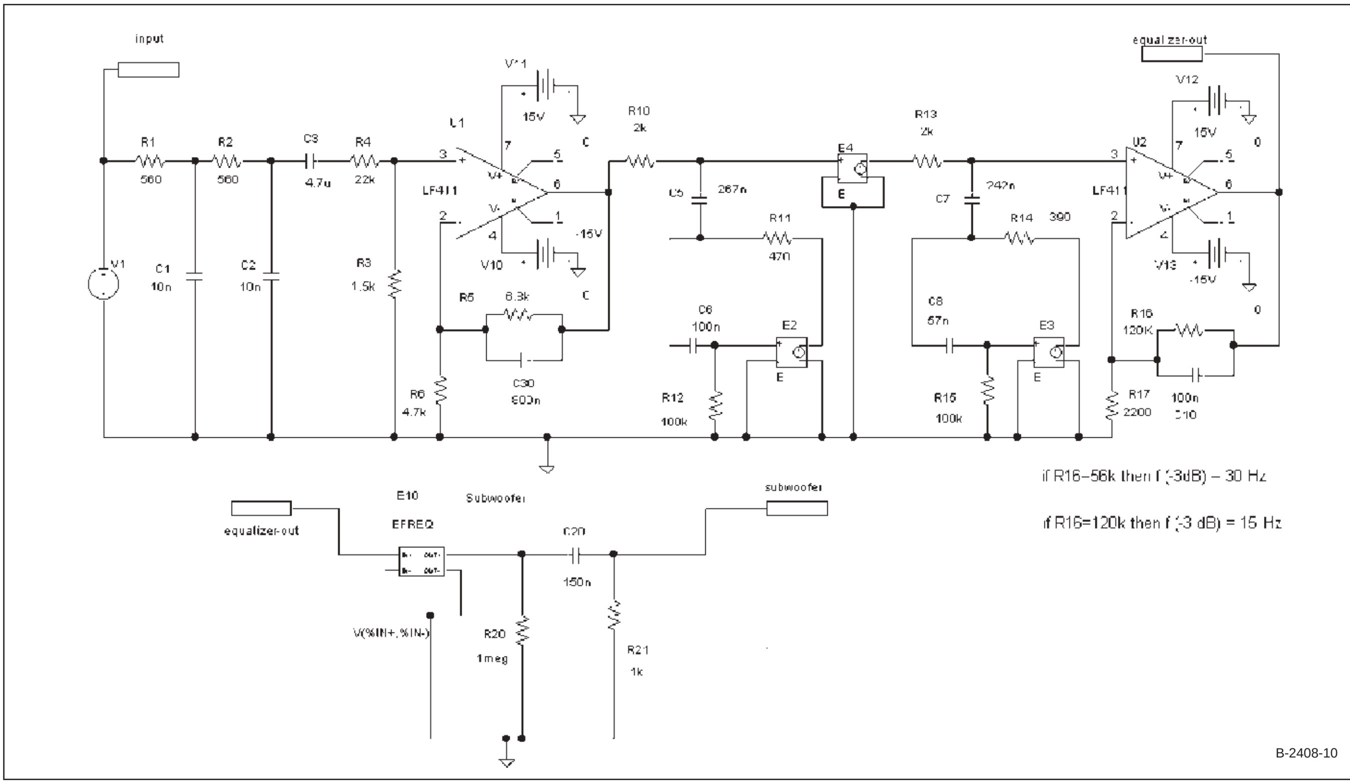


FIGURE 10: Equalizer circuit used for the PSpice analysis.

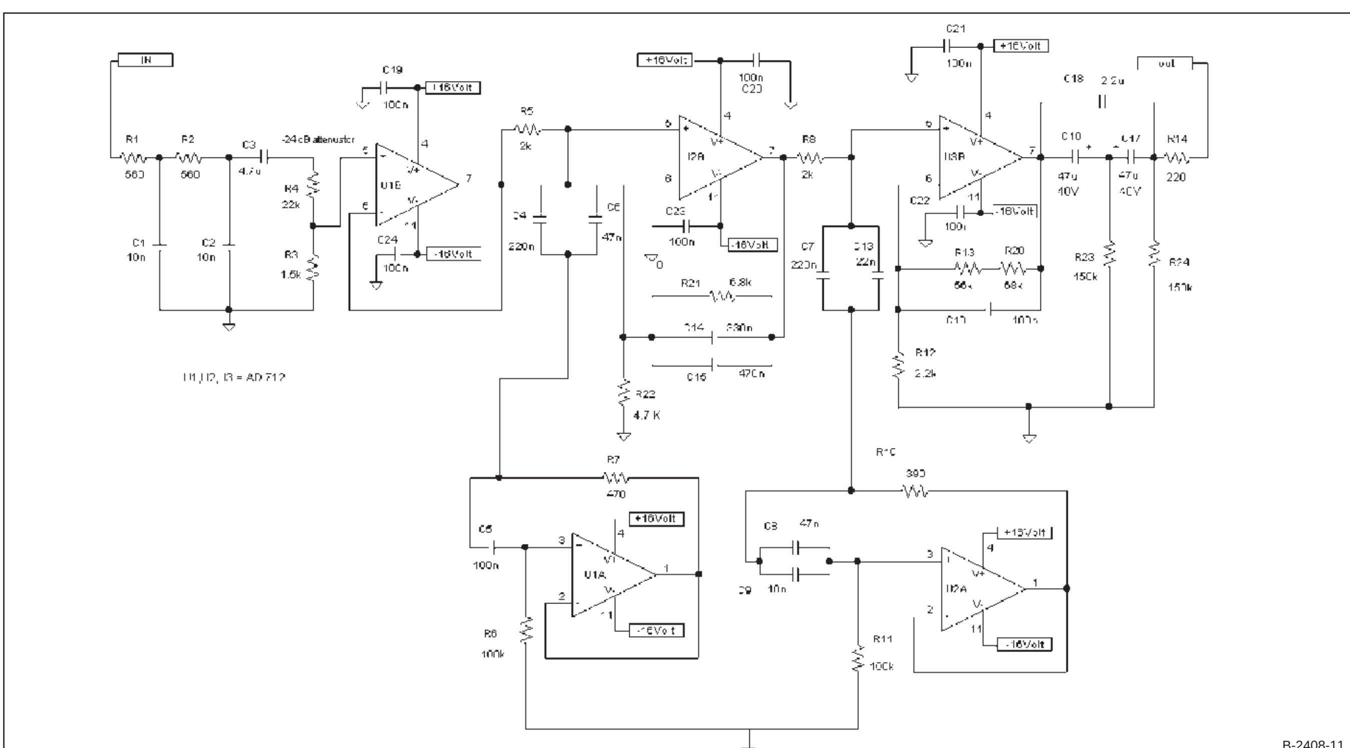


FIGURE 11: As-built electronic circuit diagram of the subwoofer equalizer.

1. This cabinet has three openings: the common, the left, and the right opening. The two drivers face one another.

The common opening of the cabinet is in the corner, while the right and left openings are at the sides of the corner. The positive radiation of the drivers comes out from the left and the right opening, while the negative radiation of both units comes out of the common opening. Mr. Linkwitz characterized this frame (on a personal communication with e-mail) as a "novel variation of the W-frame." In *Fig. 4* a top view of the cabinet is shown with detailed dimensions for construction.

This W-frame can be considered as equivalent to two dipoles at almost right angles with each other. The response at a point P that is in the far field will be equivalent to the sum of the responses of the two dipoles. Since the nominal listening axis will bear an angle with the axis of each dipole, in theory a drop in the amplitude response of the dipole will occur. Of course this is valid when the subwoofer is installed in the free field and not in the corner of a real room.

I verified this by measuring the response of the subwoofer as shown in *Fig. 5*, where the top view of the listening room is shown with the subwoofer installed in the corner. I drove the subwoofer with pink noise, band-filtered in the region between 45 and 85Hz. I used a Radio Shack SPL meter for the measurement. In all five positions where I performed the measurement, the indication was the same. This means that the directivity of the subwoofer when installed in the corner will not be a problem.

Now some important factors for the operation of the subwoofer: the distance between the two openings of opposite phase of the W-frame is not constant; it starts from 40cm and goes up to 48cm. So for the analysis, I took as distance D the mean value of 44cm. According to this theory, the peak frequency can be computed as follows:

$$Fpk = 0.5 \cdot 344/(0.44) = 391\text{Hz}$$

The frequency at which this open-baffle subwoofer has the same level output with a single monopole can be computed as follows:

$$Feq = 0.17 \cdot 344/(0.44) = 133\text{Hz}.$$

Another important factor for a subwoofer is the maximum SPL that can be produced with a given frequency. Mr. Linkwitz provides an Excel file (http://www.linkwitzlab.com/spl_max1.xls) with which the excursion limited RMS sound pressure level (SPL) for a driver in an open baffle in half space can be computed.

In my design, I used four woofers, each having Sd = 530cm^2 and Xpk = 6.75mm. I put as effective path difference D = 0.44m. Considering all the preceding, the maximum SPL that can be obtained into half space is as follows:

FREQUENCY (HZ)	MAXIMUM SPL INTO HALF SPACE WITH 4 WOOFERS (DB)
20	87
30	97
40	105
50	111
60	115
70	119
80	123
100	129
150	139

Since the subwoofer will be installed in

TABLE 1
LIST OF COMPONENTS FOR SUBWOOFER EQUALIZER CIRCUIT OF FIG. 11.

RESISTORS	
All resistors ½W, metal film	
R1, R2	560
R3	1.5k
R4	22k
R5, R8	2k
R6, R11	100k
R7	470
R10	390
R12	2.2k
R13	56k
R14	220
R20	68k
R21	6.8k
R22	4.7k
R23, R24	150k
CAPACITORS	
C1, C2	10nF, plastic film
C3	4.7m F, plastic film
C4, C7	220nF, plastic film
C5, C10, C19–24	100nF, plastic film
C6, C8	47nF, plastic film
C9	10nF, plastic film
C13	22nF, plastic film
C14	330nF, plastic film
C15	470nF, plastic film
C16, C17	47m F/40V, electrolytic
C18	2.2m F, plastic film
OP AMPS	
U1-3	AD 712

the corner of the room, I expect additional gain of several dB at the low frequencies.

THE CONSTRUCTION OF THE CABINET

For the construction of the subwoofer I used 19mm plywood for every side of the cabinet. The first step is to cut all the panels of the box according to these dimensions:

Panel A: 400 · 340mm (two pieces)
Panel B: 380 · 340mm (two pieces)
Panel C: 580 · 340mm (one piece)
Panel D: Two of these panels are needed, one for the top and another for the bottom cover. The panel has the shape of an orthogonal triangle with each side about 765mm. I used a piece of 19mm plywood with dimensions 765 · 765mm. Then I cut it into two pieces according to its diagonal, producing panel D.

When all the panels of the subwoofer were ready, I cut the mounting holes on the two panels B to place the woofers. The proper diameter of the hole for the INFINITY woofer was 274mm.

Then I cut the top and the bottom panels D as shown in *Fig. 6* to produce the common, left, and right openings of the subwoofer.

Then all the panels of the cabinet were ready for assembly. First I glued and screwed the two panels A on the top panel D. Then the panel C and last the two panels B.

I chose to leave the bottom panel D of the cabinet removable. This is a great help for possible modifications or upgrades of the system (which usually occurs).

Photo 1 shows the subwoofer as constructed up to this point. I waited overnight for the glue to dry completely. Then I applied a lot of hot glue between the joints of all internal sides to prevent possible air leaks.

Also for cosmetics reasons, I painted all the external sides of the subwoofer black using a suitable wood paint. I put the woofers on panels B as shown in *Fig. 4*; each woofer is facing the other. I did this for symmetry reasons, which I will explain later in the measurements of the subwoofer.

I used weather stripping foam tape between the woofer and the panel and screwed the woofers with M3 bolts. I also used self-adhesive weather stripping foam between the bottom panel D and the rest of the cabinet to eliminate air leaks between them.

The two drivers of each cabinet are connected electrically in series due to their low nominal 4W impedance. You can see how they are connected together and to the amplifier in *Fig. 4*. The minus (-) terminal of the left driver is connected to the positive (+) terminal of the right driver. The (+) terminal of the left driver is connected to the (-) terminal of the amplifier, and the (-) terminal of the right driver to the (+) terminal of the amplifier. I did this to maintain the absolute phase of the subwoofer with the rest of the system.

You should verify the correct connections of the drivers with a 1.5V battery. When the (+) terminal and the (-) terminal of the battery are connected to the corresponding (+) and (-) terminals of the subwoofer (marked as "subwoofer terminals" in *Fig. 4*), the left woofer should be moved toward the left opening and the right woofer toward the right opening.

You can see the complete subwoofer cabinet before the two drivers are installed and without the bottom cover in *Photo 2*. I built two such cabinets, so the whole system includes four 12† woofers.

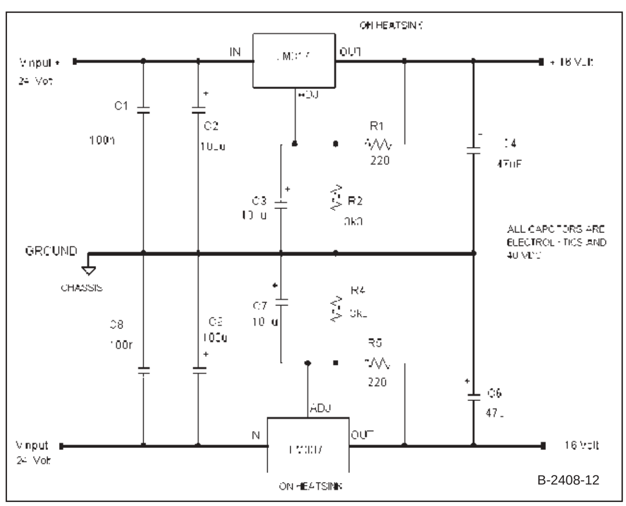


FIGURE 12: The regulated power supply.

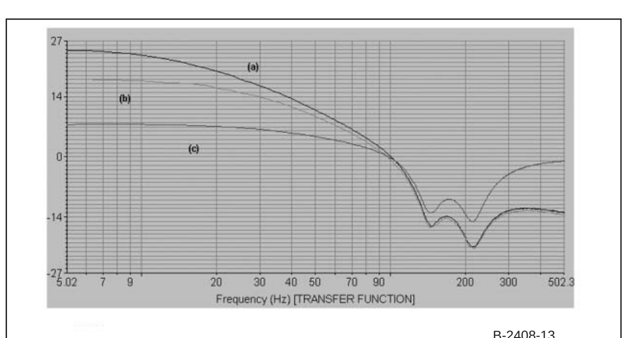


FIGURE 13: Measured response of the equalizer.

SUBWOOFER MEASUREMENT

After I completed the construction of the cabinet, the next step was to equalize the response of the subwoofer. To do this, you need to measure the near-field response.

For all the measurements I used a laptop running the JBL Smaart Pro Real Time Module version 2.1. The frequency resolution of all the measurements I took with the Smaart was 1Hz.

In the beginning, I placed the subwoofer on the floor in the middle of my listening room (position M in *Fig. 5*). It had a distance greater than 1m from any nearby obstacle. The result of these measurements is shown in *Fig. 7*.

The top line is with the microphone placed in front of the common opening of the subwoofer. The other two lines below are the near-field response with the microphone on the left and on the right opening, respectfully. (Refer to *Fig. 4* for the naming of the common, left, and right openings). The measurements on the left and right openings are very similar, proving the symmetry of the two as well as in the placement of the woofers.

The near-field response as measured at the common output of the subwoofer has some differences from the measurements on the left and right openings. The left and right openings have a broader peak response, and the frequency at which the peak occurs is different: 190Hz for the common opening and 260Hz for the other openings. This can be explained since the left and right openings of the subwoofer have different shapes from the common opening and causes the different cavity resonance frequencies. From about 30Hz to 120Hz the subwoofer's range the responses at all three openings are similar.

After this measurement, I placed the two subwoofers one above the other in the corner of the room behind the TV box (*Fig. 5*) to measure how different the near-field frequency response was from that taken in the middle of the room. The result is shown in *Fig. 8*. This response is the average of four measurements taken on the two left and the two right openings of the subwoofers.

As you can see, there are many differences between the responses when the subwoofer is installed in the middle of the room and when it is in the corner. The peak at 260Hz is still present but as expected the corner has significantly broadened the peak of the response, so now it starts from about 70Hz and goes up to 200Hz. Also, it has much more output at these frequencies actually, from 100Hz to 200Hz it has a flat response. I was not sure whether this was a problem, but I was thinking of possible ways to make the response of the subwoofers in the corner similar to the one in the middle of the room.

I found a good solution when I put the subwoofers, in the same corner, 55cm above the floor. The responses in this case are shown in *Fig. 9*. The main differences are in the range between 70 and 120Hz. The subwoofers at the new position have less output from 4dB at 90Hz to about 9dB at 110Hz.

The response is not the same as the one that I took in the middle of the room and which, I believe, has the lowest resonance, but it was the best I could have by placing the subwoofer in the corner. So I decided that this would be the final position, and I used this response as the reference for the design of the electronic notch filter.

Photo 3 shows the two subwoofers finally installed in the corner of my living room, 55cm above the floor. On the right side you can see the back projection TV that was slightly moved from its position. Below the subwoofers, there are the equalizer box, its external power supply, and the subwoofer power amplifier.

THE SUBWOOFER EQUALIZER

For the design of the equalizer, I used the Pspice 9.1 Student Version. You can download the demo version of this program directly from the Orcad website (http://www.orcad.com/downloads/demo/default. asp). This is an excellent program that can simulate the frequency response of an electronic circuit with excellent accuracy.

The demo version offers the full capabilities of the program, but the circuit must have a limited number of nodes. For this reason I replaced most of the operational amplifiers that I used as buffers with a "voltage-controller voltage source" with the Gain = 1. In PSpice,

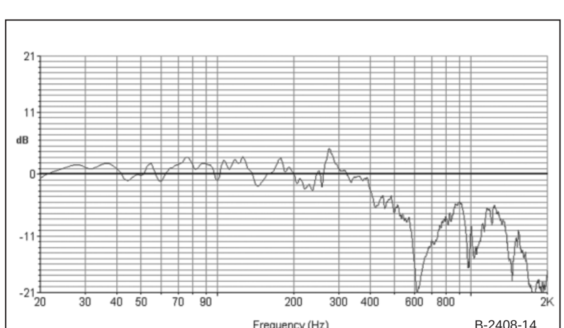


FIGURE 14: Measured response of the subwoofer after the equalization.

each operational amplifier is a sub circuit that contains many other components and additional nodes. If you used more than one or two operational amplifiers (depending, of course, on the rest of the circuit), you can easily exceed the limits of the demo program, causing it to stop running.

Another valuable capability of PSpice is the EFREQ "voltage-controlled voltage source," which you can program with a table that can have different response with frequency both in amplitude and phase. From the measured near-field response of *Fig. 9*, I produced a table from 10Hz to 400Hz with the amplitude and phase response of the subwoofer. I then inserted this table in the voltage-controlled voltage source EFREQ (component E10). After that, I changed the components' value of the equalizer on the PSpice simulator to achieve results similar to a new acoustic frequency response of the subwoofer.

This makes the development of the equalizer easier and faster. The circuit of the equalizer as used for the PSpice analysis is shown in *Fig. 10*. The final electronic diagram of the equalizer as constructed is shown in *Fig. 11*.

As you can see, both diagrams are equivalent. The differences are that in *Fig. 10* there is the source E10 that simulates the near-field frequency response of the subwoofer and the components C20 and R21; this is a high-pass filter at about 1kHz that simulates the loss of 6dB/octave at low frequencies of the subwoofer at the far field. For the analysis of the equalizer, refer to the circuit of *Fig. 11*.

By examining the circuit, you see the following main blocks:

1. The input circuit before U1B
2. The first notch filter around U1A
3. The circuit around U2B which boosts the very low frequencies around 20Hz
4. The second notch filter around U2A
5. The circuit around U3B, which boosts the 6dB/octave subwoofer frequency response
6. The output circuit around components C16-C18

The first stage with the components R1, R2, C1, and C2 comprises the low-pass input filter, which rejects the frequencies above 10kHz (remember that the circuit is used to drive a subwoofer) so as not to interfere with the rest of the circuit. The capacitor C3 blocks the DC voltage that can possibly come from the output of the previous stage that will be connected to this circuit. It has a very low -3dB point at around 1.5Hz.

The resistors R3 and R4 form a voltage attenuator to match the sensitivity of the subwoofer to the rest of my system. If this attenuation is not needed, you can replace the resistor R4 with a short circuit and the resistor R3 can become 24kW . If a different attenuation is needed, then you should set the resistor R3 to the appropriate value.

The next stage, around R5-R7, C4-C6, and U1A, is the first notch filter. It has a center frequency of 142.4Hz, an attenuation of -14.4dB, and a Q of about 8.9. For the relevant theory concerning the design of the notch filters, refer to the Linkwitz site(http://www.linkwitzlab.com/models.htm#C1).

The Appendix, which you can find at our website (www.audioXpress.com/magsdirx/aX/addenda/ index.htm), also presents a summary for the design of this notch filter. The U2B op amp buffers the output of the first notch filter and, with the components R21, R22, C14, and C15, boosts the very low frequencies so the subwoofer response will be flat to about 20Hz. It also drives the next stage.

The circuit around R8, R10, R11, C7, C8, C9, C13, and U2A is the second notch filter with a center frequency of 217Hz, an at-

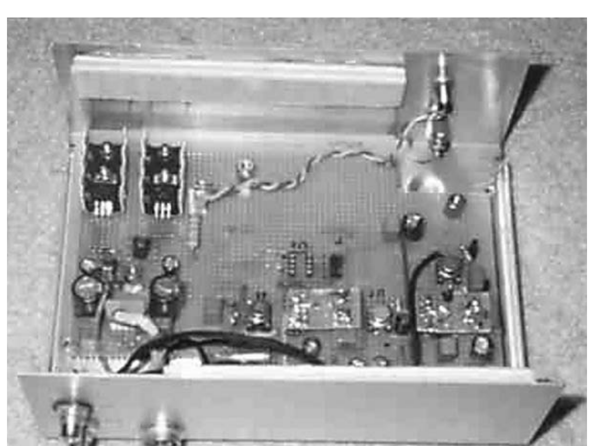


PHOTO 4: Equalizer of the subwoofer.

tenuation of -15.75dB, and a Q of 7.75. Both notch filters around U1A and U2A are needed to flatten the broad cavity resonance of the subwoofer.

The last stage around U3B is the low-frequency boost stage to equalize the subwoofer response at the far field. It has a maximum low-frequency gain of ·57 (35.2dB). It equalizes the frequency range from 10Hz to about 320Hz, and it is the driver stage for the power amplifier. Resistors R13 and R20 adjust the low-frequency response limit of the subwoofer. If R20 is short-circuited, then the limit is set to 30Hz, while if both R13 and R20 are active, the limit is set to 15Hz.

I usually set the low-frequency limit of the subwoofer to 30Hz. The output circuit around C16-C18 is the DC blocking stage for the next stage. Two electrolytic capacitors are used back to back and are bypassed with a 2.2mF film capacitor. A small resistance of 220W at the output buffers the equalizer from the cable and the power amplifier input.

The regulated power supply of the circuit (*Fig. 12*) is based on the LM317 and LM337 regulators. A small heatsink is necessary for each regulator. Resistors R1-R2 and R4-R5 adjust the output voltage at 16V.

I constructed the circuit of *Fig. 11* and the regulated power supply of *Fig. 12* on the same general-purpose epoxy PCB. I put the whole PCB in a metal box for shielding.

The DC unregulated voltage should be provided from an external power supply. You can use a common power supply provided that it has an output of about –24V DC at 500mA with low ripple.

The equalizer box with the top cover removed is shown in *Photo 4*. You can see the general-purpose PCB with the voltage regulators on the top left side and the equalizer circuit on the right side. I constructed the notch filters of the equalizer on small PCBs, which are connected to the rest of the circuit with connectors for easy removal. This was very helpful for the development of the equalizer.

After construction, I measured the frequency response of the equalizer (*Fig. 13*). Three curves are shown.

The top curve is the total response of the equalizer when the bass extension is set to 15Hz. The middle curve shows the total response with the R20 short-circuited, which sets the bass extension at 30Hz. The bottom curve is the response with the capacitor C10 short circuited that disables the 6dB/octave boosts for the low frequencies. In this case only the notch filters and the low boost circuit around U2B are active.

All these measurements were in excellent agreement with the simulated responses, and no corrective actions were needed. You can easily observe in *Fig. 13* that the difference between the low and the peak value of the response is about 45dB for the top curve and about 39dB for the middle curve from 10Hz to about 230Hz. This difference in the gain is very large; you should take great care with the construction of the circuit to avoid possible noises and disturbances.

After testing the circuit, I installed it before the power amplifier and repeated the measurements of the frequency response of the subwoofer at the near field without using the circuit around U3B (the low-frequency boost for the far-field response). I took the measurements as before at the four openings of the two subwoofers (left and right openings) when these subwoofers were placed at the corner and above the floor, and with the low-frequency limit set at 15Hz.

The average response of the four measurements is shown in *Fig. 14*.

The subwoofer near-field frequency response remains flat within –2dB between 20Hz and 350Hz. This is an excellent result if you take into consideration the fact that the frequency resolution of the measurement is 1Hz.

DIPOLE OR MONOPOLE SUBWOOFER

There is much discussion concerning the differences in the bass reproduction between a dipole and a monopole subwoofer. After a long time living with this open-baffle subwoofer, I fully agree that the dipole subwoofer is much better than the corresponding monopole type. Since I have constructed both kinds using the same drivers, I can say that the most important difference that I hear between the closed box subwoofer and the open-baffle subwoofer installed in the same listening room is more clarity in the reproduction than the dipole subwoofer.

I believe that the open-baffle subwoofer will remain in my listening room for a long time. ❖

Sonic Comparison of Power Amplifier Output vs. Input

This test methodology uses the direct approach to evaluate power amps. **By Dennis Colin**

The output of a reference power amp drives either a speaker or (through an attenuator) the input of the tested amp (*Fig. 1*), whose output then drives the speaker. An A/B switch and precise level matching allow instant comparison of the tested amp's input and output signals. Ideally, the reference amp's load in switch position B (feeding the test amp) should be a second speaker whose impedance closely matches that of the auditioned speaker.

In conventional A/B comparisons of two amps, you may prefer one over another, but that doesn't necessarily mean it's better in absolute terms (more faithful to its input source). It could have deviations from neutrality that perceptually compensate deficiencies elsewhere, or simply sound preferable to the other amp.

In contrast, the test described here allows you to hear only the colorations of the test amp. If it were the ideal "piece of wire with gain," the switching would produce no difference (other than the unproven claims of the microscopic audibility of short pieces of wire, and so on).

In *Fig. 2*, the speaker monitors the test amp's output-input difference signal (assuming a non-inverting amp). With sufficiently flat frequency response and low phase shift, you can adjust the attenuator for a deep enough null so that distortion residues can be directly heard, if sufficiently present.

A SPECIFIC TEST

The tested amp was "Mad Katy," a 125W per channel stereo unit, each channel comprised of four KT88s in push-pull Class AB with 67% screen tapping (nearly triode but 2× power) and non-feedback output stage linearizing. To (more or less) double any deviations from neutrality, the two channels were cascaded, each attenuated to unity voltage gain. The first channel was resistively loaded; the second drove the speaker.

The reference amp was a 3W single-ended EL84 unit with a high enough damping factor (15W/8Ω load) so that the changing load (8Ω versus the speaker) was apparently of no sonic significance. The speaker was the Swans M1 (reviewed in *SB* 3/99), a very high transparency system with a ribbon tweeter. Its excellent coherence, even in the near field, allowed close monitoring (1M) for optimum in-room clarity. I performed the test in mono.

RESULTS

With 12 music selections and white

noise, no audible A/B differences could be consistently—if at all—heard. I performed switching with long segments, and also with rapid repetition of the first two seconds or several notes of the piece.

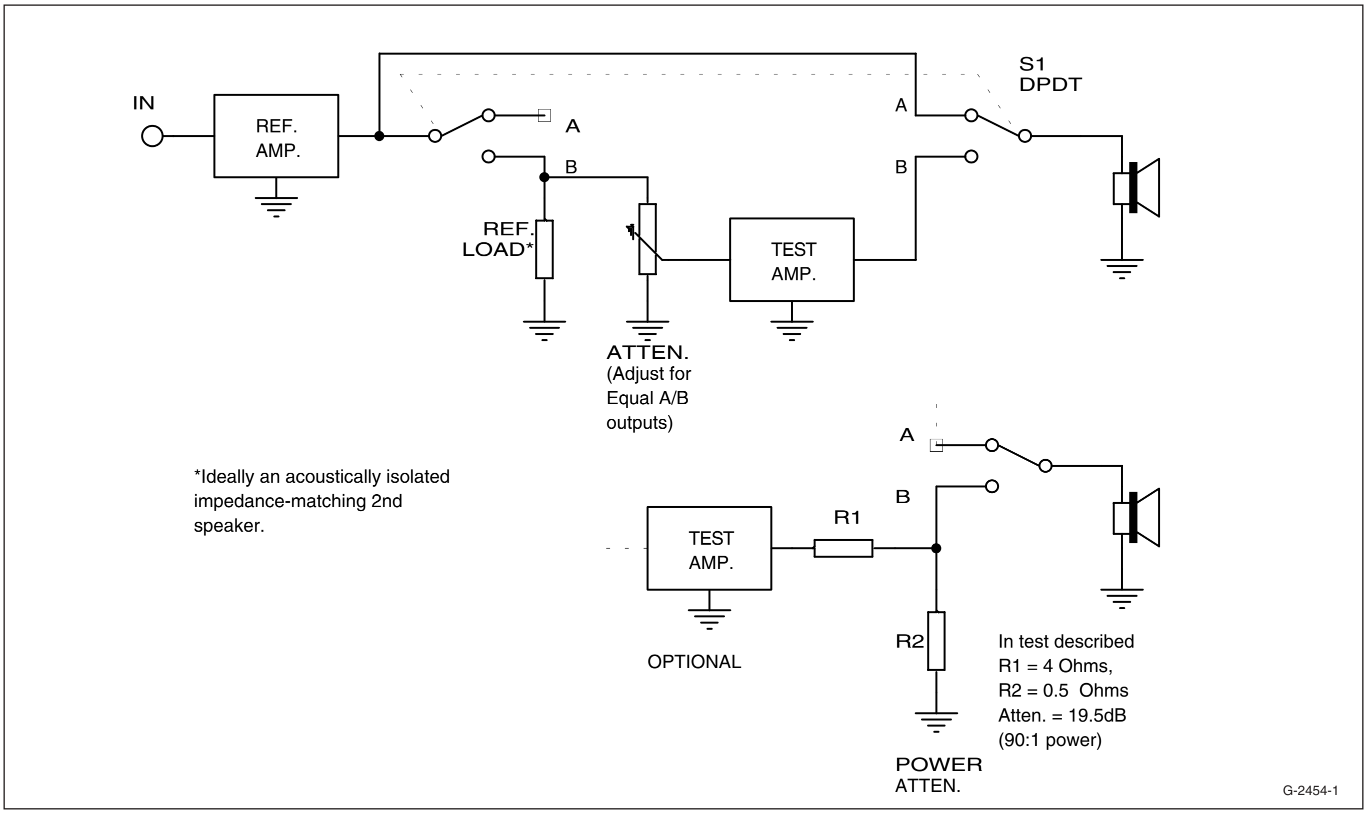


FIGURE 1: Sonic comparison of power amp output vs. input.

With the latter, the exact and frequent repetition with input/output switching was a most sensitive test. But after much switching, I concluded that the amplifier's anomalies were below audibility.

FULL POWER TEST

In the previously described test, the test amp's channels were each attenuated to unity overall gain. But because the reference amp's maximum power is about 3W, the test amp (125W per channel) output was also limited to 3W.

In a second test, a power resistor network attenuated the test amp's output by 19.5dB (90:1 power ratio). A/B levels were re-matched. Now, with 1.4W from the reference amp, the test amp was driven to 125W, with 1.4W reaching the speaker. A drawback of this method is that the power attenuator presents a nearly resistive load to the test amp output, isolating it from the varying and reactive speaker impedance. To eliminate this drawback, the reference amp would need a maximum power capacity comfortably greater than that of the test amp; this wasn't available.

In this test, admittedly with the unfairly "nice" test amp load, no audible differences could be established. This doesn't mean that the amp is "perfect," but rather that in these two tests (with two test amp channels cascaded), colorations or other changes weren't audible to me.

OUTPUT-INPUT DIFFERENCE SIGNAL

With the test setup of *Fig. 2*, I achieved a null of over 30dB. With the resulting low SPL, my sensitivity to anomalies was certainly impaired. But for what it's worth, the sound was not noticeably distorted until overdrive became (rapidly and most detectably) audible as the drive level was increased.

A NOTE ON THE TEST AMP LOADING

In a previous test with a different speaker, whose impedance drops to 3Ω from 5–20kHz, the amp produced a slight "softening" when I used its 4Ω tap. This also occurred with the Swans M1 (7Ω at 300Hz and 20kHz) when driven by the amp's 8Ω tap. I heard no "softening" with the 4Ω tap, which I used in the tests. With this amp, you should use a tap (2, 4, 8Ω) that's lower than the lowest speaker |Z| within the audio band.

CONCLUSION

These tests are simple in concept and most likely not new. I present them here to remind you that—amid all the controversy regarding "euphonic colorations" versus "musical truth," and so on—you can evaluate an audio power amplifier without comparison to another unit, but directly.

With this test, if the sound is "better" with the amp switched in, a higher "musical truth" may be *perceived*, but it is *not* "absolute signal truth." ❖

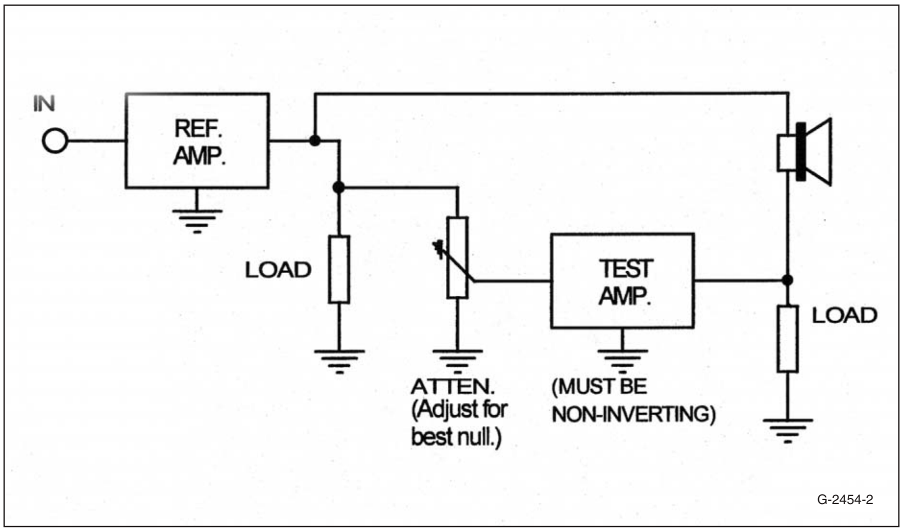
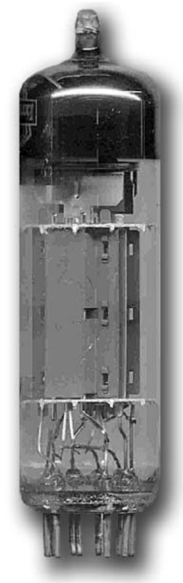


FIGURE 2: Audition of power amp output-input difference.

Why Power Tubes Arc



Much misinformation exists about the cause of arcing in power output tubes. It's time to clear the air. **By Edwin G. Pettis**

Before any tube (with the exception of cold cathode types such as 0Z4) can work properly, the heater must warm up the cathode enough to start emitting electrons which, in turn, produces a "space charge" around the cathode. This space charge actually supplies the electron current in a tube. The cathode emits electrons at a fairly even rate (this does vary a bit, instant to instant) into the space charge area surrounding the cathode.

The tube's current is drawn from this space charge at a varying rate, which is determined by the signal applied to the control grid. This, of course, varies somewhat with time, having high peaks of current taken from the space charge, which is renewed at only a relatively constant rate. Too many peak currents will begin to deplete the space charge.

CATHODE STRIPPING

When the current demanded by the signal exceeds the available electron charge, the electrons are then "pulled" directly from the cathode. This constitutes cathode stripping. When a tube is new, its cathode emissions are able to maintain the space charge sufficiently to keep up with most of the peak current demands, with only small amounts being pulled from the cathode itself resulting in little damage. As a tube ages and its cathode emission begins dropping off, the space charge can no longer keep up with current demands on peaks. This gradually builds up more and more cathode stripping, especially when the tube is called on to continually supply frequent high current peaks.

Owners who like to listen to music at very high sound levels will compensate by turning up the volume level, a self-defeating practice. This in turn causes even more cathode stripping. As this process continues, the space charge becomes more and more depleted.

This space charge has another function which rarely—if ever—is mentioned, even by so-called tube gurus. This space charge prevents arcing in a normal tube! When the space charge becomes sufficiently depleted, the tube can become susceptible to arcing.

In many cases, the owner hasn't a clue why the tube(s) arced. They may have actually seen the tube arc but cannot determine why this occurred. With no apparent malfunction of the amplifier circuitry and nothing to indicate the cause, guesswork fills in the blanks.

CHANGE-OUT

Many years ago, this causality was noted in power output tubes, but it was pointed out that this occurrence was quite rare. At that time, outside of theaters and PA systems, the average house did not place any heavy demands on output tubes—a few watts were all that were needed. The heavy users always replaced their tubes on a regular basis, which prevented the tubes from suffering premature or old-age maladies. Changing tubes regularly was cheaper than repairing amplifiers.

Today, with bigger and bigger power amplifiers, output tubes are being called upon to supply ever-increasing power levels, which they can! The problem is that these tubes will wear out faster, become unbalanced quicker, and, without due diligence on the part of the owner, will become susceptible to this type of failure mode far more often than ever thought of before.

Since most owners aren't sure when tubes should be changed, they keep using them until something happens. Either the sound quality falls enough to be noticed or tubes start arcing and failing, the latter being the rarer event. Often, even when the owner pays attention to keeping the tubes balanced, they are used beyond the point at which they should be replaced, because of the power demand put on them. The tube is still "good," even checking out okay on a tube tester, *but* its ability to provide a sufficient space charge for proper operation is declining.

You should replace these tubes before their space charge reaches a dangerous level. Tube arcing is not necessarily a result of a "bad" tube or a "bad" design. It is far more likely the result of using the tube longer than you should for the application you're using it.

An emissions-type tube tester is of some small help here, but unfortunately it cannot test for peak emission capability, so it is of somewhat limited use. One way to tell that your tubes are in need of a change-out is by how much their bias points have changed since you first put them into the amplifier. Most tubes' bias will have changed significantly as the tube ages. This can be of some help in determining when the tube is becoming weak. However, the problem is that each individual amplifier is going to be different, so that no set number can be easily assigned to "when" to change the tubes.

By the time it becomes difficult or impossible for your matched tubes to hold a "normal" match percentage, your tubes are likely already past the point at which they should have been changed. I'm not saying that every amplifier will suffer this problem or that it will occur often, but the fact is that it *can* occur often enough to cause significant damage to your amplifier. Much has to do with how and how often you use that amplifier. If it is ridden hard, it is much more likely to happen.

One thing is abundantly true: arcing is rarely ever seen in amplifiers that are *not* ridden hard; that is, called upon to supply very high power levels frequently. If you want to make those expensive output tubes last longer, turn down the volume a bit. Otherwise, be prepared to get out your wallet more frequently. It isn't just the big power amplifiers that have this problem, any amplifier can arc.

A FAIRLY COMMON EXAMPLE

Consider someone who has a pair of very expensive, high powered amplifiers, is diligent to keep bias set on them, doesn't run them too hard, but has experienced several seemingly unexplained arcing tubes, which blow plate fuse resistors (which is what they are for). In every instance, the factory said the tube was defective despite no evidence to support that conclusion. In some cases the tube that arced remained serviceable, others were damaged beyond use. In either case, a tube that has suffered an arc should not be used again.

The arc will have caused damage to the cathode whether or not a tube tester shows it. The condition that causes a tube to arc is not a defect, the tube is simply too worn out to work properly under demand. The owners who refused to believe that defective tubes were the sole cause were right. This problem has occurred both with NOS tubes—no matter the brand—and with recently made tubes. This failure mode doesn't care whose tubes are in the amplifier; they can all suffer the arcing problem given the right conditions.

Forty years ago, manufacturers such as Mullard recommended replacing an output tube when its power output fell to 50% of normal. This was fine forty years ago, when amplifiers weren't driven as they often are today. Power amps rarely were bigger than 40 or 50W, and two or four tubes could provide that much power easily.

The vast majority of amplifiers were under 20–25W, which a pair of tubes could output for a long time. A pair of KT88/6550s could put out 50–60W and have a good long life. Things have changed over the years. Many of today's owners don't have the knowledge they need to take care of their amplifiers, and neither do the manufacturers.

It is not entirely the owner's fault. Amplifier manufacturers rarely provide such information to their customers, and there is a wealth of poor or just plain wrong information on the web. Self-proclaimed tube gurus propagate much of this misinformation. Others who operate with a modicum of knowledge make assumptions based on little real knowledge and put that out on the web for others to read, and so it propagates all over, more myth becoming "fact."

It is doubtful that many tube amplifier manufacturers' technical departments are that well versed in all of the nuances of tube operation either, because little if any documentation has been printed by them addressing this problem. Since this is something of a "new" problem that has shown up in recent years in amplifiers, little thought has been given to this mode of failure.

SOLUTIONS

Because the method of turning on filaments and preheating the tubes before applying plate voltage has been observed for many, many years in transmitters, the same practice has been observed in some power amplifiers. This alleviates cathode stripping in cold tubes, but does not protect tubes from arcing during operation. This is an entirely different situation calling for different procedures. It is virtually up to the amplifier's owner to guard against the possibility of arcing, and there are no pat answers on what numbers to use. You could make educated calculations, but they would be different for each particular amplifier.

The easiest solution to this problem is to just turn the volume down and not push the tubes so hard. They will reward you with a longer lifespan and no arcing.

Cathode stripping has been addressed to some degree in publications, but cathode stripping at cold turn-on has largely been misaddressed. In the case of cathode stripping, arcing occurs only when sufficient damage has been caused by cold cathode stripping. Most amplifiers will not cause cathode stripping at turn-on. Delayed turn-on is a good idea in general but is *not* that important to tube life unless something has not been done right—a bad circuit design, for example. In fact, receiving tubes are designed to withstand higher than maximum-rated voltage when cold.

Cathode damage results only when current is drawn sufficiently before the cathode has reached a minimum operating temperature. This condition does not normally exist in audio amplifiers, even larger power amps—unless the designer has done a poor job designing it. Contrary to popular belief, slow turn-on rectifiers such as the 5AR4/GZ34 is *not* a prerequisite to safe, long-life operation of your tubes.

This "myth" was partially derived from the pre-heating requirements for high power transmitter tubes. This requirement does *not* extend to those transmitter tubes of "lower" power. The tubes' data sheet will specify whether preheating filaments is required. Popular tubes such as the 211 and 845 do not require pre-heating, however I would recommend pre-heating filaments of any tube with a plate voltage in excess of 750V as a matter of good practice. These power tubes are biased well below any danger zone for cathode stripping in audio applications. Even if the bias is derived from a tube rectifier, the bias voltage will be there soon enough to prevent any possibility of cathode stripping.

There is one exception to this situation, that is the use of solid-state rectifiers in the power supply. If solid-state rectifiers are used, it would be a very good idea to use pre-heating and delay plate voltage.

Pre-heating your amplifier's tubes isn't a bad idea, it just isn't necessary in most instances. It will not make your tubes last any longer! ❖

Room Correction, Part 4

For the final part of this room-correction system, the author shows you how he built a DSP correction preamplifier. **By Rune Aleksandersen**

PHOTO 13: The DSP correction preamplifier.

I had some thoughts about how to apply the Signal Wizard DSP-card:

- Driver crossover filter
- Direct and diffuse sound correction
- Low frequency-filter (room correction)

However, implementing all this functionality at once would require one Signal Wizard DSP-card per driver, which was not economical in my case.

There is one limitation to the Signal Wizard: it is not possible to run different filters on each channel. When working with room correction, separate filters for the left and right channels are quite important. Because of this I used the single channel mode, using one DSP card per channel. This gives me 18 bits resolution and up to 527 taps at 48kHz; 18 bits yields good dynamics (most newer DSP evaluation boards seem to use 16-bit codecs only). My application of DSP processing will be limited to direct and diffuse sound correction, essentially a loudspeaker correcting unit.

The Signal Wizard DSP cards don't come in an encapsulated version, so I decided to build the functionality into a complete preamplifier, which would contain phono preamplifier, channel switching, tape monitor output, DSP monitor output, analog or digital selector, active filter modules, and a four-deck volume control supporting high- and low-frequency outputs for a two-way system.

I decided to use parts from DACT (Danish Audio Connect[19]), as well as the two Signal Wizard DSP cards. The internal cabling was copper core foil shielded cable from Gotham Cable.

PHONO PREAMPLIFIER

I built in the DACT CT100 (in Scandinavia, it is sold as NLE17). This preamplifier has DIP switches for matching the impedance to the pickup used. In this way, any MM or MC pickup can be fitted. The buildup of the CT100 is dual mono. I chose to use the same power supply for both channels, as I think the difference of a dual or single supply for a preamplifier should be marginal.

CHANNEL SWITCHING

Channel switching is performed by the 4-pole 5-position CT3 switch from DACT. This is a very high quality ELMA rotary switch with gold-plated contacts. Being 4-pole, the switch allows simultaneous switching of both the signal path and earth. The result is that only one source is connected with the preamplifier at any time, leading to a better grounding strategy. Having tried out both the 2-pole ELMA and the 4-pole DACT versions, I am surprised how audible the difference is.

MONITOR OUTPUTS

I chose to have a tape monitor output. The output from the 4-pole switch is simply connected to a pair of RCA connectors. The outputs of the DSP cards are routed to an extra pair of RCA connectors. This makes it easy to use the DSP cards for test purposes.

DIGITAL/ANALOG SELECTOR

The Signal Wizard DSP card does not have a flash memory for storing the filter data. This means that every time the power goes down, the filter data disappears, and no sound will come through the DSP cards.

I used a 4-pole DACT switch, also switching ground, to choose either analog input from the channel switch or digital processed signals from the DSP cards. Using the selector, I can easily switch from analog to digital when doing listening comparisons.

ACTIVE FILTER MODULES

Since I am not using DSP for loudspeaker filters, I need analog filters to filter my Jordan drivers (*Figs. 93* and *94*). Active filters have many advantages compared to passive loudspeaker filters, here captured from Colloms[7]:

- Amplifiers operate over a narrower bandwidth, thus reducing intermodulation distortion.
- The amplifiers are directly connected across the terminals of each driver. The impedance seen by the amplifier is much simpler than for a passive network.
- Active filters potentially have lower distortion than passive filters, due to the elimination of inductors and the use of high-quality film capacitors.
- Active filters very easily accommodate all kinds of equalizations. Some of these are almost impossible to implement in passive networks.
- Constant delay networks that can be used for time adjustment of the drivers are easy to implement. This allows for minimum phase design.
- Sensitivity adjustments are very easy to implement by using potentiometers. For passive solutions this normally is done with a series resistor in

the high-pass filter (that causes resonance problems).

- Servo bass solutions may be implemented by including the bass driver in the feedback loop of the active filter.
- Well-made active systems subjectively have better performance than corresponding passive systems. Characterizations of "louder" and "clearer" are frequently made, and are believed due to the reduction in "stressful" loading on the individual amplifiers. For example, in a passive system, distortion in the bass will be reproduced in the treble (because the treble and bass are interconnected via the crossover). This will not occur in an active system.

VOLUME CONTROL

By using a 4-deck volume control, I have the advantage of having all signal processing, including my active filter, working at line-level signals. My old active system used an attenuator inserted into the signal path before the active filters. Usually, I don't listen very loudly (when my wife is in the room). Then, the active filters will often be working at a very low signal level, being susceptible to a bad signal-to-noise ratio as well as RFI and other interference. In general, it makes sense to amplify the signal as early as possible in the signal chain.

DSP

Each card would be configured as direct sound correction processors running at 48kHz sample rate with a 511–tap filter length. Resolution in single channel mode is 18 bits, allowing a good dynamic range.

With the DSP cards running on an internal 5V supply, this limits the input voltage that can be applied. My CD player output is around 7V running a 1kHz 0dBFS signal, so I added a pair of attenuating resistors to each channel in order that the input signal would not be clipped.

I had some challenges with EMI from the DSP cards: my receiver picked up noise in the FM band. This was much improved by adding a ferrite to the power supply cord.

POWER SUPPLY

I built a simple power supply taking a 12V AC input. The supply consists of a simple diode rectifier and three linear regulators: a positive and a negative regulator for the analog part, and a separate regulator for the DSP cards. The Signal Wizard uses 500mA per card. With 1A in my LM317 regulator, it needs a good heatsink. A suitable transformer size is 30VA or larger.

CONCLUSION

Well, how does this preamplifier sound? Compared to my old design, it sounds cleaner, and the low end of the spectrum especially is less muddled. The analog electronics clearly benefit from having the volume controls at the end of the preamplifier signal chain.

Now to the direct sound filtering: The DSP sound processing with 511 taps correcting the time window to 3–4ms makes a notable difference. First, it smoothes the spectral balance. Before, small peaks in the frequency response made some voices and instruments stand out. With DSP, this changes to a more transparent sound stage. Second, directional information in the sound stage has improved.

Before, it sounded as though some performers were simultaneously sitting in several positions. Now, stereo information seems more right. Also, the overall sound is cleaned up, making the speakers even more pleasant to listen to at higher sound levels.

Adding low-frequency correction listening to pre-filtered CDRs improves the sound stage even more. Enlightened by the results, I invested in a Behringer Ultracurve PRO DEQ2496 DSP unit for handling room correction using the parametric filters. Setting up the unit with 4ms delay made a good time adjustment compensating for the delay of the DSP preamplifier.

Despite the overall improvement of adding this digital sound processor, some dynamics are lost compared to the all-analog sound. Also, a slightly audible background hiss is present at very high sound levels. All in all, this has been a very worthwhile project! ❖

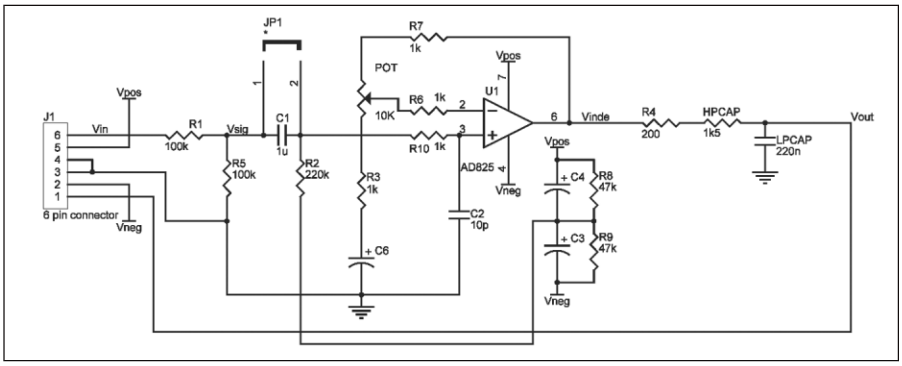


FIGURE 93: Active low-pass filter.

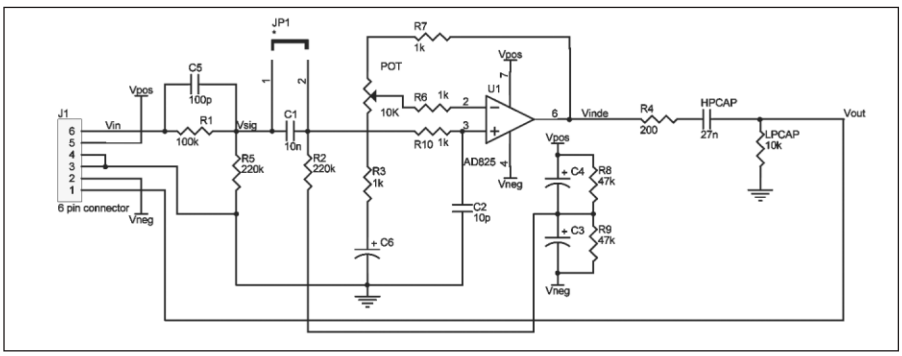


FIGURE 94: Active high-pass filter.

I would like to thank Marni Tyril for providing me with the Matlab code from the Aarhus University Master Thesis, giving this project a flying start. Thanks to Malcolm Hawksford for e-mailing me his publications. Also, the publications found on the web from Aarhus University, Anders Torger, and Angelo Farina have been invaluable.

REFERENCE:

19. DACT web site: http://dact.com/index.html.

Product Review

Pioneer DV-563A Universal A/V Player

By Charles Hansen

Pioneer Electronics (USA), PO Box 1540, Long Beach, CA 90810, www.pioneerelectronics.com, MSRP $249, (street price $180), dimensions: 16.5″W × 11.1″D × 2.2″H, net weight: 5.2 lb, warranty: 1 year.

PHOTO 1: Front view of unit.

The Pioneer DV-563A DVD player's front panel is festooned with more logos than a Formula One race car: DVD Video, DVD Audio, Compact Disc Digital Audio, Compact Disc Digital Video, SACD, MP3, Photoviewer, RW compatible, SRS TruSurround, Dolby Digital, DTS Digital Surround, and EnergyStar. While the operating manual modestly calls the 563A a DVD player, it is truly a universal audio player as well.

PHOTO 2: Remote close-up.

Although my main interests were its extensive audio capabilities, I should also mention that it has standard NTSC interlaced video output as well as a progressive scan de-interlacer chip for use with HDTV-ready television monitors. It has bass management and speaker size and distance compensation for both DVD and SACD. You can also view JPEG picture files as slideshows or in a manually selected sequence. The 563A is also compatible with DVD-RW discs recorded on a DVD recorder.

INSIDE THE PV-563A

Photo 1 shows the DV-563A front panel. The Standby/On switch is on the left. The disc tray bezel is surrounded by a number of push buttons for tray open/close, fast forward/reverse, track/chapter jump, play stop, and display dimming. The pause and play buttons are located to the right of the tray. On the right side of the front panel is a large round grouping of cursor controls and an enter button. You can use this to navigate on-screen displays and menus.

All the front panel controls except the display dimming are duplicated on the easy-to-use remote control (*Photo 2*). This light gray remote control is also easy to read, unlike those matte black remotes with gray lettering that Panasonic and Sony (among others) seem so fond of.

The front panel display has ten alphanumeric characters that help you use it in audio mode without the need for a television. The display identifies the type of audio disc (CD, SACD, DVD-A, MP3), the track number, and the remaining time—if that information is on the disc. The display also indicates when the 563A is playing the video part of a DVD, whether progressive scan is in use, when multi-angle scenes are showing, whether DTS, Dolby Digital, or TruSurround video soundtracks are playing, and whether the output has been down-mixed from the original audio source. Legends above the characters indicate DVD-Audio group numbers, DVD titles, CD/DVD track numbers, DVD chapter numbers, when analog 5.1 channel output is selected, and whether repeat play mode is selected.

The 563A is quite light, being constructed of light-gauge steel with a plastic front panel. The player base sits on four rubber squares, which may prevent any marring of your equipment

PHOTO 3: Rear view of unit.

rack, but will do nothing for vibration isolation.

The left side of the rear panel (*Photo 3*) has control in/out jacks for passing signals among other Pioneer components. Next come the optical output jack and S-video jack. There are 12 RCA output jacks for coax digital, composite video, component video, two-channel analog audio, and 5.1 channel analog audio for connection to an A/V receiver with 5.1 analog inputs.

There is no on-board DTS/Dolby Digital analog output. Full-resolution FireWire DVD-Audio and SACD digital output signals for a video processor/receiver are also not available. You would need to pay at least three times more for the former feature and six times more for the latter. A small polarized two-prong power receptacle located on the right side accepts the 2-18 gauge power cord.

On the far left side of the unit (out of sight in the photo) is a small 7-pin Molex style connector of unknown function.

Photo 4 shows the amplifier with the cover removed. The CD/DVD drive is located front and center. The phenolic PC board on the left is the switching power supply. A line fuse is located on the circuit board close to the AC line receptacle. A PC board connector with numerous gray wires distributes the various DC voltages to the rest of the unit.

The rear multilayer epoxy PC board contains the digital-analog converters (DACs) and analog output op amps. The right side multilayer board contains the optical disc controls, the DVD and SACD processor chips, and system memory. It connects to the rear board, the drive and the front control, and display PC board (located out of view under the front panel) by means of ribbon cables. All three of these boards make extensive use of surface-mount components.

TOPOLOGY

A schematic was not furnished with the unit, but a service manual is available. The 563A has a decent digital-side audio/video chip set. If there is any compromise, it is in the power supply and the audio op amp selection (fairly typical of mass-market consumer audio products).

Starting at the data source, an ST-Microelectronics STm6316 DVD Optical Disc IC handles control of the CD/DVD disc drive. It features DVD-video, DVD-R/RW and DVD+R/RW playback decoding up to 2×, and video-CD up to 6×. It supports CD-DA, CD, CD-R, CD-RW, DVD-Audio, and SACD audio playback. The IC provides full servo digital processing, twin laser support, acquisition error correction, and analog servo processing.

The largest IC on the right-front PC board is the ST-Microelectronics STi5508 OMEGA (one chip multimedia engine architecture). This integrates all of the front-end functions for linear pulse code modulation (PCM) DVD playback, including analog preprocessing, channel decoding, and error correction.

The next largest chip is the Philips SAA7893 SACD processor. SACD Scarlet Book direct stream digital (DSD) uses a sampling frequency of 2.8224MHz, or 64 times higher than that of Red Book CD (64fs). The chip

PHOTO 4: Interior view.

incorporates a multi-channel bass management system with volume control and a delay function for home theater support. The MPEG video decoder is a Sharp 033EZ01 chip. The crystal-controlled master clock runs at 27MHz. This universal audio/video system also requires 64MB of SDRAM.

The rear I/O PC board uses two different TI/Burr-Brown digital-to-analog converters (DAC). The first is a DSD1791, which provides stereo digital-to-analog converters for the front left and right channels. It supports up to 24-bit linear PCM audio data format (for CD and DVD-Audio) at sampling rates of up to

LISTENING AUDITION

In the audio modes, the DV-563A will play DVD-R/RW and DVD+R/RW audio discs, SACDs, standard CD and CD-R/CD-RW discs, and CD-ROMs containing MP3 files[3]. I also tried to play 24/96 AIFF files and 16-bit WMA files, but the 563A rejected those, just as its operating manual said it would, displaying "STOP" followed by "NO PLAY."

When you insert a disc, the unit determines the format before playback begins. This takes some time, given the large number of formats the 563A is capable of, and the fact that all shiny silver 120mm discs initially look the same. I measured the time it took from the "LOADING" display until the format was recognized and accepted for play in each mode:

SACD	6 seconds
DVD–A	5 to 17 seconds until MENU appears (depends on content)
DVD+R	20 seconds
CD	8 seconds
CD-R	18 seconds
CD-RW	20 seconds
MP3	15 seconds
DVD–V	14 seconds until MENU appears
JPEG	14 seconds

Track jumps were fast in SACD and CD modes, but fairly slow with DVD. Another difference was that the audio could be heard while fast-forwarding SACD and CD discs, but not during DVD-Audio. Fast-forward and fast-reverse were very leisurely in all the audio modes.

Bass management varies depending on the audio source. The –12dB/octave subwoofer output low-pass filter characteristics vary from 100Hz (DVD-video and CD), to 120Hz for SACD, to 200Hz for DVD-Audio. These filter turnover frequencies are fixed and cannot be changed to match your actual speaker characteristics. Hum and noise were practically non-existent with my ear to the speakers with a –90.31dB 1kHz audio CD test track playing to prevent output muting.

The music video tracks that I viewed on my TV were quite well done. Since I do not have a DTS/Dolby Digital processor, I did not evaluate the Group 3 DVD-video tracks for audio quality. Once you introduce the video aspect of the performance into the picture (no pun intended), the perceived inadequacies of the audio tend to be less objectionable. The final DVD track group includes an artist biography, an audio interview, a photo gallery, and a catalog. DVD jewel cases are larger than those for CDs and SACDs in both depth and width, and won't fit in standard CD/SACD rack storage slots.

I began my listening audition with selections from the standard *audioXpress* audition tracks on the *HFN&RR* CD, comparing the 563A with my modified Rotel RCD 970BX CD player[4].

The massed voices in *Jerusalem/Parry* were noticeably clearer and more distinct with the Rotel, with better imaging and a wide soundstage. The Rotel has a more extended high-frequency range and brighter midrange than the Pioneer does.

The brasses and strings in Vivaldi's *Trumpet Concerto* were clear and bright with both players, with no dissonant overtones. The harpsichord in Vivaldi, in *Henry V Extract/Doyle*, and in *Welcome, Welcome/Purcell* was detailed and in balance with the other instruments in the Rotel. The Pioneer began to lose it as the music became more complex.

The electric bass in the *Rio Napo* track had good bass extension with both players, but the Pioneer was muddy and tubby in the lowest registers.

I continued with other CDs from my collection. I found imaging to be a bit less precise with the Pioneer, and it didn't have near the depth of the Rotel. During low-level fadeouts the Rotel could go further into the decreasing audio level before the sound turned grainy. The "woody" tone of an acoustic bass was better defined with the Rotel, while the Pioneer's upper bass was a bit louder.

Overall, the Rotel was easier to listen to and more revealing, while the Pioneer was a bit more veiled and laid back. The CD performance of the DV-563A didn't have any major vices such as harshness and glare; it just wasn't as involving or as pleasant to listen to as the Rotel over an extended period of time.

Since the DV-563A can also play MP3 files, I burned some selected CD tracks to 128K-sampled compressed MP3 files with the Creative PlayCenter CD ripping software[5]. My perception was that the MP3 tracks were a bit louder than the CD tracks of the same songs. Imaging was not very distinct and the instruments on some recordings had a hollow, synthetic sound. The soundstage was flat with no depth whatsoever, and any reverb or ambience that was present in the CD version was lost in the MP3. This actually improved the sound in some pop music songs where various effects were applied to excess in the original CD tracks.

Next I turned my attention to the 563A's SACD performance. I began with the dual-layer 5.1 multichannel SACD by Norah Jones, "come away with me" (Blue Note 7243 5 41747 2 8). My NHT SW-2P subwoofer has a separate amplifier whose input is derived from my two-channel preamp output, so the fixed LP subwoofer output filters in the 563A were not a factor. I did my two-channel SACD listening with the 563A menu set to large speakers and with the subwoofer on.

Compared to the Rotel playing the CD version of this same disc, the Pioneer had a more pleasant presentation throughout (I had no other SACD player available with which to compare the Pioneer's performance). The piano was more distinct and realistic, and the drum kit had better definition than with the CD on the Rotel. The

192kHz. It also supports 44.1kHz × 64 sampling for the 1-bit DSD audio data format that SACD discs use.

The DSD1791 provides balanced voltage outputs. There are a number of through-hole film capacitors whose values are consistent with those needed for the analog audio post-filtering output stage specified in the DSD1791 data sheet. A dual op amp is mounted on the bottom side of the PC board to implement the active filter. I could not determine the op amp type (TI/Burr-Brown recommends the OPA2134 or 5532).

Two DSD1702 DACs are used for the center channel, surround channels, and subwoofer outputs. These stereo DACs also support both LPCM and

Pioneer's overall response, imaging, and depth were more uniform in SACD mode than in its CD mode. I think Pioneer did a fairly nice implementation of SACD in the 563A player.

I continued auditioning other SACD tracks from the Chesky "An Introduction to SACD" (SACD204) with the same pleasant surprise (more on this later, in the DVD-Audio audition).

High-definition DVD-Audio discs are in short supply. The primary aim of DVD seems to be the video/home theater market. I chose a DVD-A disc that had all the various flavors of DVD-Audio: Two-channel 24-bit/96kHz, 24-bit/96kHz surround sound, and Dolby Digital surround sound as well as Music Video tracks: the Diana Krall DVD "When I Look In Your Eyes" (Verve B0001286-19). I also have the CD version (Verve IMPD-304), which I played for comparison on my Rotel player.

If I merely selected PLAY with the DVD in the tray, the player defaulted to Group 1, the advanced resolution surround-sound tracks. These were rather dreadful sounding on a two-channel stereo system, so you knew right away that you had the wrong DVD group. The guitar was left somewhere out in the hall, except for brief chords where it jumped back to the left front. The high end of the keyboard had a similar treatment on the right. The drum kit was still in the front-center, but lost some resolution compared to the 24/96k DVD stereo. I suspect the tracks in the surround-sound version are in 16/48k or 20/48k format.

Unfortunately, I had to again make use of the video setup menus to select song groups and tracks on the DVD. I connected the composite video output of the 563A to the composite video input of an old Sony VCR. Then I ran a long RG/6 video cable from the VCR modulator F-video output to the TV in the next room so I could map the cursor settings for each DVD mode. Why can't DVD-Audio players have a button or local display selection for "2-channel audio" so you don't need to get involved with a TV. It would even be easier if the track numbers didn't start from track 1 again as you change groups. Again I set the menu for large speakers and subwoofer on, and selected the Group 2 two-channel 24/96 tracks.

The Pioneer's DVD-Audio 24/96 performance was not as impressive as it was in SACD mode. When compared to the same CD tracks playing on my Rotel, the drum kit was not as well defined and the acoustic bass was not voiced in a realistic woody manner. The strings and horns sounded a bit better than on the CD, and Miss Krall's voice sounded more "in the room." The overall sound was a bit less congested during loud passages of massed instrumentation than with the CD. The fadeout endings were clear all the way down to the noise floor. However, the performance was not strikingly better overall when compared to the CD played on the Rotel.

Next, I sampled the Chesky DVD-A "The John Basile Quartet—The Desmond Project" (CHDVD178), for which I have the Chesky CD version (JD156). Basile's guitar and Allen Mezquida's alto sax are a bit more involving on the DVD than on the CD, but there is still a problem with the presentation of the acoustic bass. The hint of reverb on the DVD recording is more delicate than on

DSD audio data formats. Each DSD1702 includes an 8× digital interpolation filter for PCM signals.

A digital DSD filter provides three different selectable frequency response options, followed by enhanced multi-level delta-sigma modulator employing 4th-order noise shaping and eight-level amplitude quantization. Analog audio post-filtering and output buffering for the center, surround, and subwoofer outputs is handled by a pair of Rohm BA4560 dual op amps. The four LP filters use 5% chip resistors and chip ceramic caps.

The 4560 op amp was derived from the old Raytheon RC4558 first-generation low-noise bipolar op amp topology

the CD.

I continued my audition with 24/96 DVD-A tracks from the Chesky DVDs "The Ultimate DVD Surround Sampler and 5.1 Setup Disc" (CHDVD221), and "The Super Audio Collection & Professional Test Disc, Part I" (CHDVD171), reaching the same conclusions about DVD-Audio. There are tracks on both the Chesky SACD and DVD discs by five Chesky artists (Rebecca Pidgeon, Livingston Taylor, the Conga Kings, Carla Lother, and David Chesky) and one song that is common to both formats (*Well I've Been to Memphis*, by David Johansen and the Harry Smiths). This allows for a one-to-one comparison between the two high-def formats.

Both discs are well recorded in the Chesky tradition. In my judgment, the SACD playback produced tighter bass, better depth and definition, and a bit wider soundstage. (This comparison would have been easier if it weren't for the slow disc access times and the need to access menus on the DVD-A disc.)

The Chesky DVD-Audio recordings were slightly better than the CD layer of the Chesky SACD disc played on my modified Rotel. There was a noticeable improvement in the Wagner DVD-A track *Tannhäuser: Act II, Scene IV–Chorus*. The massed choral and orchestra were nicely defined and the imaging was accurate.

This may be due to the lack of deep bass in this classical piece. The lack of depth when compared with SACD was still noticeable, however. I don't know whether Chesky uses dedicated DVD and DSD recorders for mastering their DVDs and SACDs, or makes a super bit-mapping direct (SBMP) conversion from SACD to LPCM (or vice versa) for the two different catalog formats.

My final audition of DVD-Audio was the AIX Records "DVD-Audio/Video Start Here" disc, which contains a few tracks recorded in 24-bit/192kHz DVD-A format. This is a double-sided disc, and fortunately it wasn't too thick for the Pioneer drive tray. The "blue" side is recorded in DVD-video. The music tracks have been mixed in both "stage" and "audience" perspectives for comparative purposes, and you can switch between the two with the remote control Audio button.

(You must sit through the FBI warning on the DVD-video side even though there is no "video" per se—just stills from the album covers and the menus. The player won't let you fast-forward through the FBI warning, displaying "STILL" on the screen while you are forced to read at a pace that would bore the average second grader). From the vagueness in the imaging I noticed using the two-channel setup mode, they seem to have been mixed for DTS and/or Dolby Digital surround. On to the real audio tracks!

The "red" side of the disc contains the DVD-A selections. The menu system on the AIX disc was different from that on the Chesky disc, which was different again from the Verve disc. This meant yet another trip to the TV to map all the button pushes that were required to get from group to group. You would think that the track jump buttons would take you sequentially through the chapters and groups, but it doesn't.

Once you reach the last track in a group, you needed to address (circa 1975), with 10MHz GBW product and a limited slew rate of just 2.8V/μsec. Its input stage dynamic range[1] (V_{th}) is only 0.04V. This is not the type of device used in high-end audio components, and much better audio op amps are available. Since these limited-bandwidth channels will not be active for stereo listening, this may not be a major problem assuming a high-quality dual op amp is used for the front L and R channels.

the menu again. This made it totally frustrating when I tried to compare the Laurence Juber *Cannery Row* guitar track, which is recorded in both 24/96 and 24/192 DVD-A formats. Between the long track access times and the need to use the menus, it was impossible to make a fast A-B comparison. I had to stop concentrating on the music and try to remember the remote button combination to get from one group to the next.

The 24/96 tracks on the DVD-A side are in 5.1 Meridian Lossless Packing (MLP) format. I previously set the DV-563A up for two-channel, large speakers. As before, my preamp provides the two-channel output for my subwoofer amplifier, so I did not become involved with the subwoofer output of the player. The 24/192 tracks are stereo only, in both PCM and MLP format. Unfortunately, there are no complex orchestral or choral pieces in 24/192. The tracks are limited to small acoustic group performances and two solos.

Starting with the AIX 24/96 tracks, I found their recording quality nearly equal to the Chesky DVD-A disc. The three classical orchestral pieces were more two-dimensional than I would have expected from high-resolution DVD-A.

The pianissimo sections were detailed enough, but not very involving. The small chamber pieces were a bit nicer. Woodwinds were bright and clear and I could identify the flute, clarinet, oboe and bassoon when all were playing simultaneously, even with the still noticeable lack of depth. The reeds weren't quite "reedy" enough for my taste, however. The imaging, sound stage, and depth were much better with the SACD discs.

The jazz selections on the AIX disc were well recorded, but the Pioneer's muddy bass problem showed itself again. Percussions, chimes, triangles, brushed snare drums, and cymbals were presented quite well, by comparison. The sax and trumpet solos on the Jim Dejulio track were not brassy enough for my taste.

Unfortunately, as I mentioned earlier, there was only one duplicate song between the 24/96 and 24/192 tracks. Menuing back and forth between the two showed only a barely discernible difference in favor of the higher sample rate. As if to mock my attempts to evaluate 24/96 vs. 24/192, the entire selection of 24/96 DVD-A format songs are duplicated on the DVD-video (blue) side of this disc.

After a fairly exhaustive search, I found very few DVD-A discs that have been released in 24/192 format. Most of these are re-mixed rock and pop music from the '70s and '80s. AIX has a fair number of classical and jazz releases (I'm not sure of the DVD-A format they used), and they shipped me the "Start Here" disc in only three days. They produce only DVD-A format products.

DIGITAL OUTPUT

In order to evaluate the digital output of the DV-563A, I connected its coax digital output to the digital input of the Alesis ML-9600 Masterlink, using the ML-9600 DAC and analog section for playback.

Playing CDs on the 563A produced the expected 16/44.1k data stream at the ML-9600. The sound was noticeably improved over CD playback on the 563A. Since both machines use modern 24-bit DACs (TI/Burr-Brown and AKM, respectively) I attribute the improvement to the better analog section in the ML-9600. As a CD player, the modified Rotel has a slight edge over the ML-9600, but that's a whole 'nother review (see the ML-9600 review in *aX* 4/04, page 46).

MP3 file playback also produced a 16/44.1k digital transfer, with the same reduced bandwidth I observed on the 563A. The ML-9600 has noticeably better imaging and depth than the 563A, making it a more involving listening experience.

DVD-Audio file digital transfers were decimated from 24/96k down to 16/48k, as required by the DVD specification. Piping the 563A DVD playback through the ML-9600 got rid of the tubby bass problem, while giving up a bit of depth, definition, and soundstage as compared with the high-def 24/96k. I believe there was a better overall balance to DVD-Audio through the ML-9600, despite the loss of bits and sample rate. The biggest tip-off that this was not 24/96k was the grainy nature at the end-of-song fadeouts.

Both Red Book CDs and DVDs have a 0dBfs output level specification of 2.0V RMS. The extra eight bits are used to extend the theoretical "analog zero" from –96.33dBfs down to –144.49dBfs. The noise floor of the DAC and analog stages usually limits the effective dynamic range to 20 bits (–120.41dBfs). The DV-563A has an effective dynamic range specification of 108dB, or about 18 bits.

The single-bit SACD format samples DSD at 2.8224MHz. It uses aggressive low-frequency noise shaping to remove 1-bit Gaussian sampling noise up to 100kHz, and is also said to approximate 20-bit performance. SACD playback did not produce any digital output, since it's incompatible with the S/PDIF coax PCM output format.

All the op amps are capacitive coupled with United Chemi-Con aluminum caps and run on a single 11.5V DC supply. Muting transistors are located adjacent to each analog output jack.

The video processor is an LA73054 IC. A single 78M05 linear regulator IC

CONCLUSION AFTER AUDITION AND TEST

The Pioneer DV-563A has received rave reviews in the consumer A/V magazines, and has a significant following in the DIYaudio forum for audio-related modifications. I don't have a full-blown home theater system, but these publications have praised its video performance as above average. Most DVD players—universal or not—tend to have their design emphasis on the video. The 563A is easy to use in its many modes of operation, although the DVD modes (even two-channel DVD-A) require the use of the on-screen menus. The operating manual is thorough, clear, and has sufficient handholding detail for even the most technically challenged buyer.

Although universal in its capabilities, the DV-563A is anything but uniform in its playback of the various audio formats. The audio deficiencies I found during the listening tests are ones of omission rather than commission. CD playback and the audio on DVD-video performances are equal to many mid-fi players, so I wouldn't buy the DV-563A if my sole interest in music were CDs.

While it's adequate as a DVD-A player, the SACD playback was better than I expected from such a reasonably priced unit. I would be willing to have it in my listening room to demonstrate SACD to those who are interested. Since all the audio formats share the same DACs and op amps, it's possible that the less engaging PCM format performance lies with the first-generation PCM processor chip used in the unit. However, this seems to be refuted by the noticeable improvement in sound when the Pioneer's PCM digital output was played through the DAC and analog stages of the ML-9600. The 563A just might be a nice low-cost candidate for a power supply and analog op amp POOGE project.

For the most part, DVD-A seems to have conceded the highest resolution audio market to SACD. DVD-A's tight integration to those video menus makes it maddening as a two-channel audio format. As of this writing there is still no DVD-Audio equivalent of the dual-layer SACD disc. A CD/DVD-Audio flip disc (CD on one side and DVD-Audio on the other) continues to be "imminent." —CH provides regulated 5V DC for all the DAC chips. (Some high-end players have a dedicated regulator IC for each chip in the unit.)

There are a number of large unused surface-mount IC patterns on the rear board, presumably to allow for additional functions in more expensive DVD players in the Pioneer and Elite lines of products. The DV-563A does not have a Pacific Micronics PMD200 filter/decoder chip and is not HDCD compatible.

MEASUREMENTS

After several days of listening auditions (see sidebar), I performed measurements on the DV-563A using the following test discs. I did not attempt to measure the Dolby Digital or DTS audio performances.

- Sheffield/A2TB "My Disc," CBS Labs CD-1 and Pierre Verany test CDs.
- CD test tracks that I ripped to 128KB sampled MP3 and burned to CD-R.
- Philips Super Audio CD DAC Test Disc.
- Chesky Super Audio Collection & Professional Test Disc, Part II (despite the name, this is a DVD-A disc).
- Analog test tracks that I recorded in 24/96 AIFF format and had burned to DVD+R (on a Mac using Digidesign ProTools|HD with a TDM system).

I put the DV-563A in CD playback with a CD test disc for one hour before making any measurements. The player was cool to the touch over its entire surface after this period.

The output impedance, for both 2-channel and the 5.1-channel outputs, measured 460Ω at 20Hz, 440Ω at 1kHz, and 436Ω at 20kHz. There was 872Ω DC resistance between the two left front jacks (stereo and front-channel 5.1), indicating they are driven from the same op amp with an isolation resistor from the op amp to each jack. I found the same resistance between the two right front jacks. L-R channel balance was 0.02dB. The L and R surround channels had a bit higher imbalance, at 0.2dB. The S/PDIF digital coax output impedance measured the specified 75Ω.

The analog outputs had normal polarity, a positive-going test pulse producing a positive-going output. The digital black test tracks measured –104dB, indicating the muting transistors were probably shunting the outputs during this test. Front left and right channel separation measured –110dB from 100Hz to 10kHz, with both unweighted and A-weighting filters. The channel separation test tracks consist of a 0dB sine wave on one channel, and digital black on the opposite channel whose crosstalk is being measured. The DV-563A appears to mute the analog output of the silent channel during this test. The center and surround channels showed about 10dB higher residual noise levels than the front channels.

In CD mode, the player performed perfectly in the track defect dropout tests out to track 36 on the Pierre Verany test disc, which contains a 2.5mm gap (Red Book requirement is 0.2mm). Even at the largest 4mm defect, it produced only occasional clicking in the audio output.

The 1kHz 0dBfs CD output at the front channels was 2.15V RMS at 1kHz,

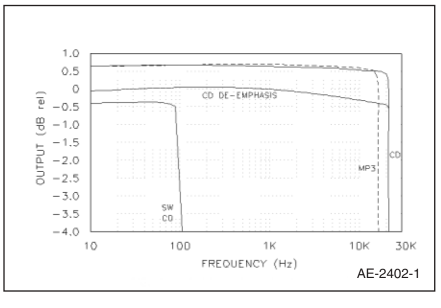


FIGURE 1: Frequency response (CD and MP3).

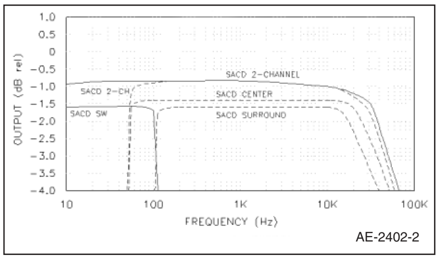


FIGURE 2: Frequency response (SACD).

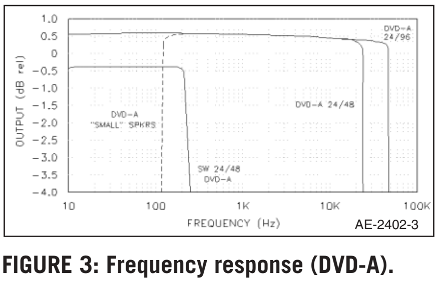


FIGURE 3: Frequency response (DVD-A).

or 0.63dB higher than the CD Red Book standard of 2V RMS. Frequency response for CD and MP3 is shown in *Fig. 1*. There was about 0.5dB of de-emphasis response error above 11kHz when playing the pre-emphasized audio test tracks. The MP3 frequency response with identical CD test tracks ripped to 128k MP3 format was 0.04dB higher through the midband before dropping off just above 15kHz. The subwoofer output was –0.37dB below that of the front channels before rolling off at 100Hz.

The higher definition 24/96 DVD-Audio and SACD shows the extended frequency response of which these formats are capable. With two-channel SACD playback, the 1kHz 0dBfs test track produced a level of 1.81V RMS, or –0.87dB below the Scarlet Book standard (*Fig. 2*). When I selected SACD 5.1 mode, small speakers, subwoofer on, and the default fixed output levels, the front L and R tracks remained at 1.81V RMS.

The center output was 1.72V RMS and the surround and subwoofer outputs were 1.68V RMS (subwoofer measured at 50Hz). The front channels rolled off just above 60kHz, while the center and surround channels rolled off below 50kHz. This is consistent with the analog output LP filter stages recommended by TI/Burr-Brown for the DSD1791 and DSD1702 DACs, respectively.

Setup for the DVD-Audio measurements required a trip back to TV land to map the on-screen menus for the test tracks I needed. With either a 24-bit/48kHz (24/48) or 24/96 1kHz 0dBfs DVD-A sine wave, the output was 2.13V RMS, or +0.55dB compared with the 2V RMS requirement. The subwoofer output using 24/48 0dB at 100Hz was 1.91V RMS (*Fig. 3*).

As you can see in *Figs. 1–3*, the –12dB/octave subwoofer low-pass filter turnover frequency depends on the selected audio mode: 100Hz (DVD-video and CD), 120Hz for SACD, and 200Hz for DVD-Audio. The latter is too high for clean reproduction by larger subwoofers, and would undoubtedly produce high bass imaging problems with a single subwoofer.

THD+N versus frequency is shown in *Fig. 4* for each audio mode. I engaged the 22kHz LP filter in my distortion test set. The SACD test signals have such high out-of-band HF signal energy that I had to build an additional ninth-order passive 20kHz pre-filter[2] to prevent slew rate limiting in the active filter stages of my distortion test set.

Note that the SACD test tracks are recorded at –3dBfs, and the DVD-A test tracks are recorded at –6dBfs. This presents a problem in displaying the data since THD increases as the output level drops. I normalized all the distortion curves to the 0dBfs output level of the CD/MP3 tracks by adjusting them to the THD I measured for the 1kHz 0dBfs tracks recorded on the SACD and DVD-A test discs.

THD+N versus output voltage is shown in *Fig. 5*, at 1kHz for each audio mode. The subwoofer THD was measured at 50Hz.

The residual distortion signal in CD mode for 1kHz at 0dBfs (*Fig. 6*) consisted mainly of very low-level noise. The MP3 residual distortion signal (not shown) was similar, with a bit higher noise level.

It has become my practice, when a piece of equipment uses a switching power supply, to examine the spectrum extended to 166kHz for any evidence of the power-supply switching frequency in the analog output. In *Fig. 7* you can see noise bursts up to –66dB around 77kHz (a fairly common personal computer power-supply switching frequency). There are also –78dB spikes at 22kHz, –81dB at 44kHz, and –79dB at 96kHz and 104kHz. I wonder whether these are artifacts of the clocking required for the various audio formats.

During this test I played a 1kHz test track at –90.31dB to prevent output muting. While 77kHz is well outside the audible range, the frequency response available with SACD and 24/192 DVD-Audio goes out to 96kHz. This should not present a problem as long as power-supply-related intermodulation products do not intrude into the audio band.

The distortion residual for the SACD playback of a 1kHz sine wave shows a higher residual noise level as well as noticeable HF "fuzz" on the sine wave itself (*Fig. 8*). The residual noise, when viewed on a wideband analog oscillo-

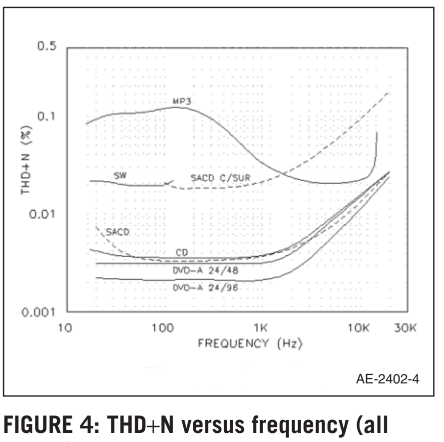


FIGURE 4: THD+N versus frequency (all modes).

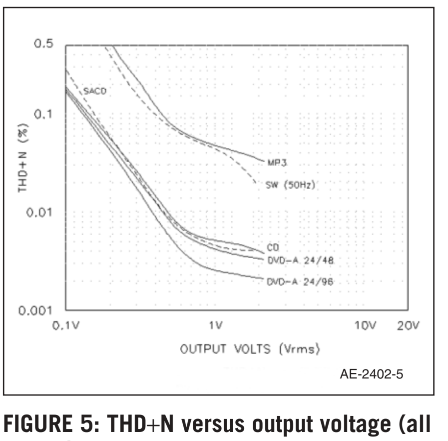


FIGURE 5: THD+N versus output voltage (all modes).

scope, has a much higher frequency content than is shown here in the DSO capture.

This is a product of the one-bit technology used in DSD. The delta-sigma DACs alter the noise spectrum of the audio signal, so the SACD processor uses aggressive LF noise shaping to move the noise levels up beyond the audio band, increasing the out-of-band noise. A conventional second- or third-order analog filter at the output of the DAC then limits the HF noise that is produced at the analog output.

You can see this effect in the extended spectrum analysis of the 1kHz –90dB SACD test sine wave with the passive 20kHz LP pre-filter removed (*Fig. 9*). The noise floor increases significantly above 20kHz, high enough to hide the power supply spikes that were visible in the CD playback spectrum (*Fig. 7*). The HF noise levels out, or shelves, above the DAC analog LP filter breakpoint.

The SACD noise shaping should produce a roughly constant downward slope in the noise floor below the HF shelving point. However, the DV-563A SACD output seems to hit a LF noise floor that is higher than the noise floor seen in the PCM modes (CD and DVD-A). This higher noise floor is reflected in the distortion measurements. The SACD audio band THD+N measures 0.034%, but the FFT computation of the THD alone is 0.0038%.

(This noise floor behavior is better illustrated by a ⅓ octave analysis, but my DSO cannot display this type of spectrum. Its 16-bit ADC sampling is also challenged by SACD and 24-bit DVD-A high-resolution audio, which typically demonstrates 20-bit or better dynamic range. I can compensate somewhat with low-noise preamplification of these low-level signals, but I cannot really achieve the performance of a true

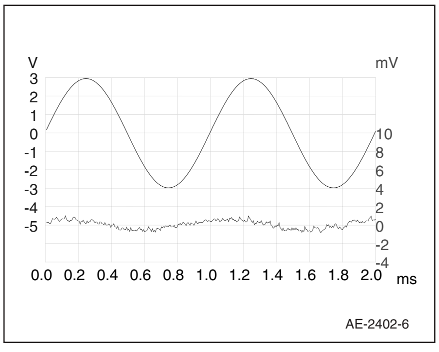


FIGURE 6: Distortion residual of 1kHz sine wave (CD).

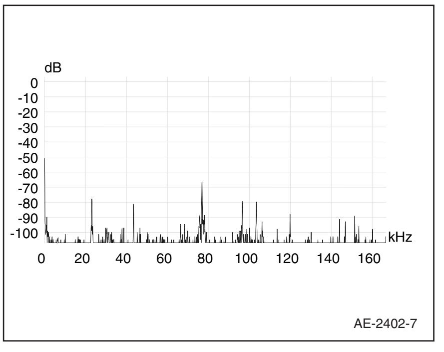


FIGURE 7: Spectrum of 1kHz at –90.31dB extended to 166kHz (CD).

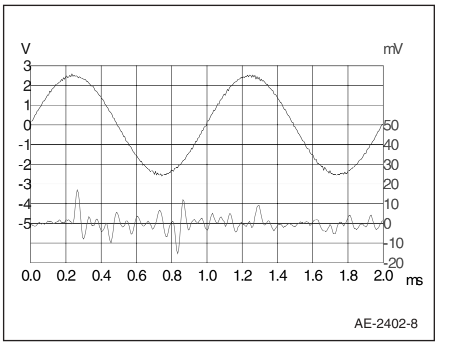


FIGURE 8: Distortion residual of 1kHz sine wave (SACD).

24-bit ADC.)

The residual distortion signal in DVD-A mode with 24/96 data for 1kHz at 0dBfs (*Fig. 10*) consists mainly of very low-level noise. The residual noise level for 24/48 data (not shown) is a bit higher, but less than that for the CD mode. The extended spectrum of a 1kHz –90dB DVD-A 24/96 sine wave has an exceptionally low noise floor (*Fig. 11*). While the power-supply-related spikes are present, there is no evidence of what appeared to be the clock-related spikes I found in the CD spectrum (*Fig. 7*). Reducing the sample rate to 24/48 (also not shown) brought up the noise floor, but otherwise the spectrum looked the same.

For the spectrum of a CD 50Hz sine wave at 0dBfs from DC to 1.3kHz, visit www.audioXpress.com. The calculated THD+N based solely on harmonics was 0.0039%, which is the same value measured with the distortion test set. It stayed at this level through 1kHz before rising at higher frequencies. This shows that for CD playback, very little noise is contributing to the total THD+N. The second harmonic measured –103dB, the third was –90dB, the fourth was –101dB, and the fifth was –93dB. There are no 60Hz power-line harmonics or other spuria evident in the spectrum.

The MP3 spectrum shows a different story (for spectra and undithered sine waves, visit www.audioXpress.com). The 50Hz fundamental sits on a broad band of noise out to 300Hz or so, with additional spikes at 750Hz (–77dB) and 1250Hz (–64dB). The THD+N measures 0.11%, although it drops to 0.034% at 1kHz.

With the passive 20kHz LP pre-filter installed, there is about 5dB less noise in the SACD spectrum of 50Hz than there is when reproduced from the CD. The FFT-calculated THD is 0.0039%. However, there is a train of spikes at multiples of 60Hz all across the spectrum. With a little experimentation, I traced these spikes to magnetic field pickup in the four air-core inductors in the pre-filter. You can see the effect of this in the spectrum of a 1kHz sine wave at 0dBfs, where it results in a band of 60Hz harmonics out to about 4kHz. The DV-563A is not producing these 60Hz artifacts.

The DVD-A test disc did not have 50Hz test tracks, so I used the 1kHz test signals at 0dBfs. The spectrum analysis to 20.3kHz with 24/48 DVD-A data yielded a THD+N of 0.0033%. The FFT-calculated distortion was 0.003% with second, third, and fourth harmonics at –100dB, –92dB, –119dB, respectively.

Increasing the DVD sample rate to 24/96 produced a much cleaner spectrum. The THD+N measures 0.0022%, which is essentially the noise floor of my distortion test set. The FFT-calculated THD based solely on harmonics is 0.002%, with the second harmonic at –105dB, the third at –95dB, and the fourth at –104dB.

You can see the CD spectrum of response to equal level 11kHz and 12kHz signals, each at –6dBfs, from DC to 20.8kHz. The 1kHz intermodulation distortion (IMD) difference product measures a low –95dB. The 10kHz product is –88dB and the 13kHz product is higher at –81dB. There is another spike near 20kHz.

When the same IMD signals are ripped to MP3 format, the results are not as good. The 1kHz product is still a low –94dB, but the 10kHz and 13kHz products have moved up to –76dB. The skirts around the 11kHz and 12kHz stimulus signals are much

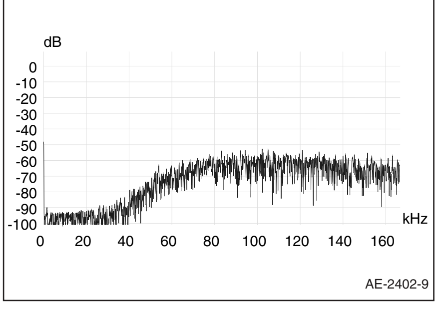


FIGURE 9: Spectrum analysis of 1kHz –90dB sine wave extended to 166kHz (SACD).

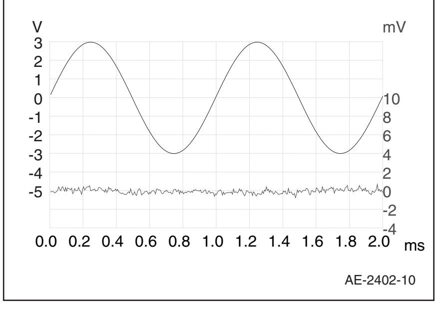


FIGURE 10: Distortion residual of 1kHz sine wave (DVD-A at 24/96).

broader, indicating more overall noise. There are also unrelated spikes of about –75dB at 6.5kHz, 7.5kHz, 8.5kHz, and the previously mentioned spike near 20kHz.

A repeat of the test in CD mode with the more difficult 19kHz + 20kHz IMD test track shows the 1kHz intermodulation difference product to be –88dB (0.004%). The 18kHz product is below –90dB. This is commendable performance.

It is impossible to discern the 1kHz IMD product in the SACD reproduction of the 19kHz + 20kHz intermodulation test signal due to the passive pre-filter related noise level. However, the 18kHz product is below –100dBfs.

You can see the DVD-A spectrum of response to the 19kHz + 20kHz IMD signals using 24/96 test data, from DC to 20.8kHz. The 1kHz intermodulation distortion (IMD) difference product measures –88dB, the same as the CD test. The 19kHz product is –92dB and there is a spike at 2kHz of –88dB. Using 11kHz and 12kHz IMD signals and DVD-A 24/48 data (not shown) produced essentially the same results as the CD test at those frequencies. This seems to point to the analog stages as the limiting factor in IMD performance.

At the level of the undithered 1kHz sine wave at –90.31dBfs, the signal consists of ±1 bit of data, producing two different voltage levels that are symmetrical about the horizontal axis (time). These discrete voltage steps, while not perfect, are recognizable.

The same CD track ripped and played in MP3 mode shows what appears to be a better-looking sine wave. I think that noise and compression artifacts have obliterated the vertical transitions in the output voltage, making it more like a sawtooth wave.

Noise also took its toll on the reproduction of the –90dBfs SACD test track at 1kHz. You can just make out the undulations of the sine wave.

A repeat of the –90dBfs signal level from the 24/48 DVD-A test disc shows the much better resolution that high-definition audio formats bring. There are now 9 LSB bits of data making up the sine wave. Extending the sample rate to 24/96 DVD-A improves the sine wave by virtue of the lower noise floor. With an A-weighting filter engaged, the signal-noise ratio was 120dB.

The CD playback of a 0dBfs square wave at 997Hz (*Fig. 12*) exhibits the Gibbs phenomenon ringing associated with the steep digital filters used in the DV-563A. This ringing is supposed to be a damped sinusoid, but the ripple appears to be clipped off at the positive and negative peaks. Perhaps this indicates that the maximum 3.1Vpp analog output voltage of the DSD1791 DAC

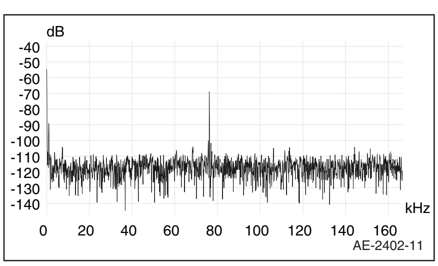


FIGURE 11: Spectrum analysis of 1kHz –90dB sine wave extended to 166kHz (DVD-A at 24/96).

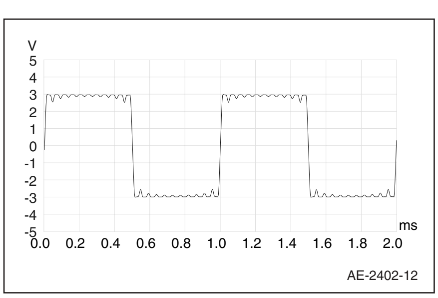


FIGURE 12: 997Hz square wave response (CD).

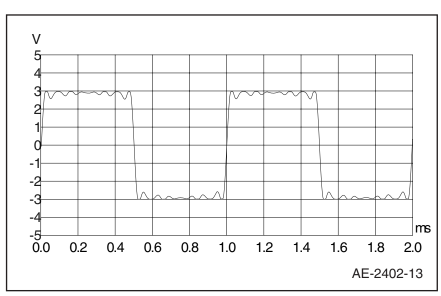


FIGURE 13: 997Hz square wave response (MP3).

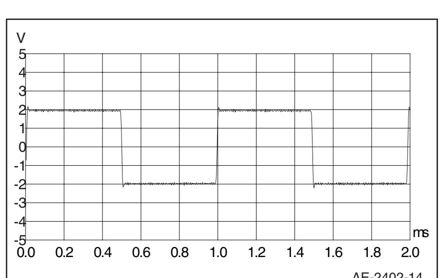


FIGURE 14: 1kHz square wave response (SACD).

causes the op amp to reach its output swing limit.

The 0dBfs sine wave does not quite reach this level: 2.15V RMS has a peak level of 3.04V. No clipping is evident in the 1kHz sine wave in *Fig. 6*, or in the 50Hz spectrum. If this were a problem with the anti-aliasing filter in the DAC, it would show up as a reduced overshoot ripple at the trailing edges of the square wave.

Reproduction of the same square wave ripped to MP3 showed similar clipping (*Fig. 13*). Note that the Gibbs

TABLE 1

MEASURED PERFORMANCE PARAMETER	MANUFACTURER'S RATING	MEASURED RESULTS
Component Video Out (three RCA jacks)	Y (luminance), 1Vpp (75Ω) P_B, P_R (color), 0.7Vpp	
S-Video Out (S-Video jack)	Y (luminance), 1Vpp (75Ω) C (color), 0.7Vpp	
Composite Video Out (RCA jack, yellow)	1Vpp (75Ω)	
Audio Out (2-channel CD) compared to Red Book standard	200mV (1kHz, –20dB) L, R	2.15V RMS (1kHz, 0dBfs), +0.63dB
Audio Out (2-channel SACD) compared to Scarlet Book standard	200mV (1kHz, –20dB) L, R	1.81V RMS (1kHz, 0dBfs), –0.87dB
Audio Out (2-channel DVD-A)	200mV (1kHz, –20dB) L, R	2.13V RMS (1kHz, 0dBfs), +0.55dB
Audio Out (5.1 multi-channel), six RCA jacks	200mV (1kHz, –20dB) fixed levels SACD (all channels) (1kHz, 0dBfs)	SACD 1.81V RMS
DVD-A	DVD-A	
L and R Front	2.13V RMS (1kHz, 0dBfs)	
Center	1.72V RMS (1kHz, 0dBfs)	
L and R Surround	1.68V RMS (1kHz, 0dBfs)	
Subwoofer	1.68V RMS (50Hz, 0dBfs)	
FREQUENCY RESPONSE (–3DB):		
CD fs:44.1kHz	4Hz–20kHz	4Hz–20kHz
MP3	N/A	4Hz–15kHz
DVD-Audio fs:48kHz	N/A	4Hz–23kHz
DVD-Audio fs:96kHz	4Hz–44kHz	4Hz–47kHz
DVD-Audio fs:192kHz	4Hz–88kHz	(test tracks not available)
SACD	N/A	4Hz–62kHz
SACD center/ surround channels	N/A	4Hz–47kHz
S/N ratio	118dB	120dB, 1kHz, "A" weighted, DVD-A 24/96
Dynamic range	108dB (approx. 18-bit)	
Total Harmonic Distortion	0.0014%	0.0039% 0dB 1kHz CD 0.034% 0dB 1kHz MP3 0.0038% 0dB 1kHz SACD (see text) 0.0022% 0dB 1kHz DVD-A 24/96
Wow and flutter	±0.001% W. Peak or lower	
Output Impedance, Analog	N/S	440Ω, 1kHz
Optical Digital Output	Toslink	
Coaxial Digital Output	SPDI/F RCA	
Output Impedance, Digital	75Ω	75Ω
Power Requirements:	14W maximum, 0.12W Stand-by	

pre and post echo ripple has a total of 16 "pulses" per full cycle, compared with the 22 pulse CD playback. You can approximate the PCM Nyquist frequency response limit by multiplying the number of pulses by the fundamental square wave frequency. This demonstrates the more limited high-frequency response available as a result of the MP3 compression algorithm.

This clipping is not evident in the leading edge peaks of the SACD 1kHz square wave output at 0dBfs in *Fig. 14*. Recall that the SACD 0dB output level is 1.5dB lower than that of the PCM outputs in the DV-563A. I don't believe the Gibbs phenomenon occurs with the DAC operating in the SACD mode.

SACD reproduction does not require the steep digital filters needed for PCM conversion to analog. The SACD DAC outputs are rolled off with an external low-order 77kHz analog filter. This is what shelves the rising HF noise in *Fig. 9*. Note the visible noise "fuzz" on the horizontal portions of the SACD square wave.

The playback of a 1kHz 0dB square wave in 24/48 DVD-A mode (*Fig. 15*) shows the 24 pulse Gibbs ripple associated with the higher resolution 24-bit DAC performance. There is again visible clipping on the peaks of the ripple, as there was with the CD and MP3 square waves. Since the DVD-A test disc has a digital filter characteristics test track (1kHz square wave at –6dBfs), I used it to further explore the Gibbs ripple clipping. *Figure 16* shows the response, where the Gibbs ringing clipping has disappeared.

Increasing the sample rate to 24/96 1kHz 0dBfs square wave data again shows clipping on the 48 pulse Gibbs ripple (*Fig. 17*). Decreasing the square wave to –6dBfs produced the clipping-free waveform in *Fig. 18*. ❖

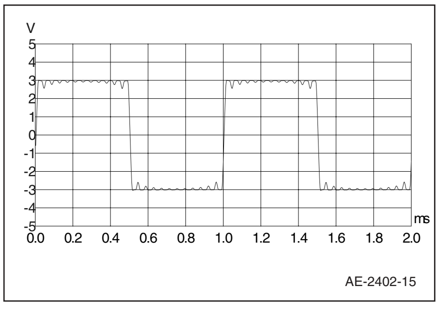


FIGURE 15: 1kHz square wave response (DVD-A 24/48).

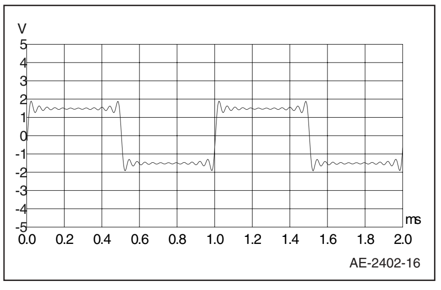


FIGURE 16: 1kHz square wave response (DVD-A 24/48 –6dBfs).

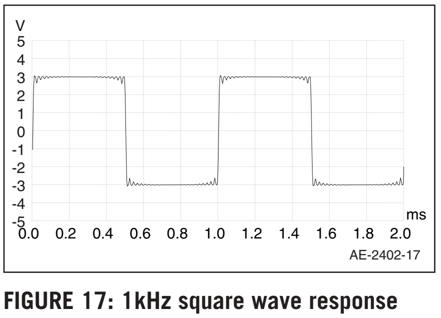


FIGURE 17: 1kHz square wave response (DVD-A 24/96).

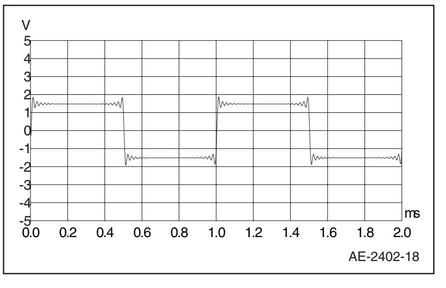


FIGURE 18: 1kHz square wave response (DVD-A 24/96 –6dBfs).

REFERENCES

1. "Op Amp Meets CD," Walt Jung, *TAA* 3/86, pp. 7–9.

2. The AES17-1998 specification calls for a steep low-pass filter for THD+N measurements for DACs that exhibit high levels of out-of-band noise. The filter specifications call for a 20kHz passband with >60dB attenuation from 24kHz to 200kHz. I designed my passive filter with a 600Ω input impedance and 10k output impedance. I wound the air core inductors with 23-gauge HTZ magnet wire on 3″ OD coil forms, and used polypropylene film capacitors. I used a filter design program called Filter Solutions from Nuhertz software, see www.filter-solutions.com

3. For a more detailed discussion of the various digital audio formats, refer to "Digital Audio Breaks the Sound Barrier," Brian Dipert, *EDN* magazine pp. 71–90, July 20, 2000.

4. "Upgrade Rotel's 970BX CD Player," Hansen, C., *audioXpress* Feb. '03, pp. 26–31.

5. The various types of compressed audio encoders have improved quite a bit over the years. For a more detailed description and extensive tests, see "Digital Audio Gets an Audition," Brian Dipert, *EDN* magazine, pp. 87-106, Jan. 18, 2001; and "Facing the CODEC Challenge," David Ranada, *Sound and Vision*, pp. 98–100, June 2002.

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www.audiokits.com
Selected Borbely Audio kits in Japan:
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AD INDEX

Xpress Mail

AUDIO INTERCONNECTS

In their review of audio interconnects (July 2004), the authors spent considerable time and effort evaluating four types of cables for their sonic attributes.

In the case of car audio, susceptibility to noise pickup (alternator whine) is the most important consideration for interconnects. In the home, noise susceptibility is less important, but there are still sources of interference such as nearby televisions, computers, and so on. Mr. Hansen mentions the need for shielding against electromagnetic interference, but doesn't include this factor in the review.

There have been reviews of audio interconnects comparing the noise susceptibility of different cable constructions such as coax and unshielded-twisted-pair (UTP), but none with the unbiased technical expertise shown by Mr. Hansen. Such an evaluation is tricky, because there is no standard source and load impedance, and some automotive equipment uses differential inputs to break ground loops. But if Mr. Hansen is so inclined, a followup review of this nature would be a valuable contribution to readers. Reference: Navone, David, and Richard Clark, "Misconceptions in Cabling," AUTOMEDIA 2, no. 6, 1997, 29–42.

Mark Rumreich
Indianapolis, IN

Charles Hansen responds:

Noise susceptibility in the home environment is equally important since the dB noise level in a car is much higher than in the home, so noise pickup at home would be much more noticeable. Even a luxury sedan has a background sound level of about 65dB at cruising speed due to wind and road noise.

There are three potential sources of electromagnetic interference (EMI) in car alternators. The first is the noise due to the full-wave three-phase rectification of the alternator stator AC voltage. A large capacitor is installed on the diode heatsink, which, in conjunction with the inductive reactance of the stator windings, forms a very effective low-pass EMI filter.

The second source of EMI is the integral solid-state voltage regulator, which uses a power transistor to provide pulse-width modulated (PWM) current to the alternator field. This module also incorporates LP filtering.

The third source is the brush noise from the field slip rings. These brushes are used to de-

liver the voltage regulator field current to the rotating field windings. Since the slip rings are a smooth surface rather than the series of bars and insulators in the commutator of a brush-type DC motor, this is a minor factor.

As part of a program instituted by NASA in the early 1990s to investigate the use of automotive components in private aircraft, I ran extensive testing on a Ford alternator from 2000 rpm (idle) to 12,000 rpm at various load conditions, and with various battery conditions from new to near-dead. We noted the lack of real radio interference problems with modern alternators. I could detect no alternator "whine" in the factory audio system in my own car with my ear to the A-pillar mounted tweeters.

Because of the proliferation of microprocessors in modern automobiles, car manufacturers expend a huge amount of engineering and development to ensure that the control systems and sensors in their vehicles are not susceptible to EMI sources, including the alternator and other onboard electrical equipment. They also must ensure the vehicle is not a source of interference. Ground loops have the potential (no pun intended) for all kinds of electrical mischief due to the wide use of frame ground connections that eliminate many of the ground return wires. Some of the systems that use control computers are the engine control module, automatic transmission control module, anti-lock brakes (ABS), traction and stability control, anti-theft/security, climate control, active suspension, and so on.

I pulled the service manual electrical diagrams for my car and found that the following sensors have shielded wiring:

- *L&R knock sensors*
- *L&R oxygen sensors*
- *Primary and secondary crankshaft position sensors*
- *Primary and secondary throttle position sensors*
- *Mass airflow sensor*
- *Fuel pump control module*
- *ABS/TCS wheel speed sensors*
- *Six airbag deployment accelerometers*
- *Line-level signals from the dash head unit to the door and rear audio amplifier units*

Reputable car audio system installers use proven methods when installing aftermarket audio equipment to ensure compatibility with the vehicle control and safety equipment. Ham radio transmitter installations are among the most difficult due to the high RF field generated from the transmitting antenna. The American Radio Relay League (ARRL) publishes an excellent book called Radio Frequency Interference: How to Find It and Fix It. *See their website at arrl.org.*

HELP WANTED

This is an open letter to all the electronic designers reading *audioXpress*. There must be thousands of readers who still own a turntable. It has been well established that improving the power to your turntable motor improves the sound. There are a number of power conditioners for synchronous turntable motors on the market—all of them prohibitively expensive.

Isn't there someone who is capable of designing such a power supply so *audioXpress* readers may build it and improve the sound of their vinyl? Such a power supply should have a clean sinus output and some sort of lock for both 33⅓ rpm and for 45 rpm and enough power to drive a turntable motor.

Ernie Brunings
brunings@sr.net

My father was a ham and electronics repairman for over 40 years. Unfortunately, he passed away a few months back. Now we have piles of old tube-type equipment and two large boxes of miscellaneous vacuum tubes. We are not sure of the value of anything.

I know some of the tubes are brand new, never used, and there are other tubes that I am sure are dead or have seen many hours of use. We have found local ham contacts for most of the equipment, but we have no clue what to do with the tubes. We remember dad saying that some of the tubes are very expensive, but we have no idea which are expensive, cheap, good, or bad. Can you give us any advice on how to resolve this? The equipment is located just outside of Nashville, Tenn.

Kevin Longfield
Research Director
NOP Automotive
klongfield@nopworld.com

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

Line Stage Odyssey Continues

From a simple transformer-output line stage to a world-class, state-of-the-art, differential parafeed design. **By David Davenport**

It all started back in February and March 2003 *aX* ("Odyssey of a Line Stage") with a simple transformer loaded, single-ended-triode amplifier that has long been a part of the audio designer's catalog of circuits. It is renowned for stable performance and excellent drive capability. Part of the attraction of the circuit shown in *Fig. 1* is its simplicity.

CIRCUIT DESCRIPTIONS

In this circuit everything is in series. All current flowing through the tube also flows through the transformer and the power supply with the tube acting as a valve to modulate this current. The modulated current induces a magnetic field in the transformer, which in turn provides current to the output. The DC component of the current develops a voltage across resistor Rk that is used to bias the tube to the desired operating point. Capacitor Ck provides a low-impedance path for the signal so that it will bypass Rk and thus not modulate the voltage produced by Rk.

Although this circuit works very well, it does have a couple of shortcomings when applied to high-end audio. Take a look at the signal current for a moment.

Of course, the signal must flow through the tube and the transformer in order to produce an output, but unfortunately it also flows through the power supply and Ck. In fact, Ck is only there to bypass the signal current around Rk. Ck usually needs to be a very large electrolytic capacitor so that it can pass very low frequency signals. The audible effect of electrolytic capacitors is well documented, and it is desirable to eliminate them if at all possible.

The signal current also passes through the power supply, and what is usually in a power supply? More electrolytic capacitors, and perhaps other stuff that may also hurt the sound.

Now consider the power-supply current, which must flow through the power supply as well as the tube and Rk. However, it serves no purpose for the transformer, and in fact may be detrimental for optimum performance. So it would seem to be a good idea to separate the signal current from the power-supply current. In fact, this is one of my criteria for a high-fidelity amplifier. *Figure 2* shows one way that you could accomplish this.

This circuit provides two separate parallel paths for the signal current and the power-supply current. The only component common to both paths is the triode. This is called a parallel feed, or "parafeed," circuit.

Capacitor Cp provides a low-impedance path for the signal to flow while blocking any direct current. Because of the impedances in this circuit compared to those in the cathode circuit, this capacitor can be several orders-of-magnitude smaller than Ck, and thus may be a high-quality film capacitor. You may select a transformer that is not designed to carry any direct current for use in this circuit. This is important because when a transformer doesn't need

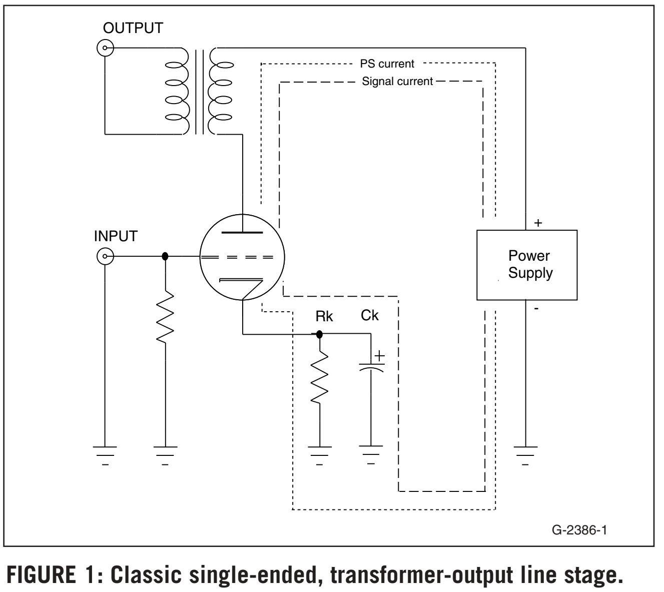


FIGURE 1: Classic single-ended, transformer-output line stage.

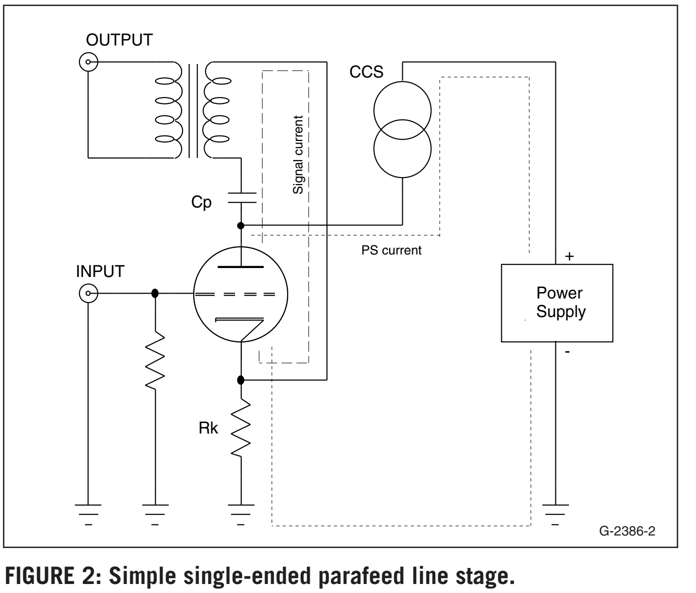


FIGURE 2: Simple single-ended parafeed line stage.

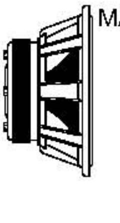
Hi-Vi Research Loudspeakers

For complete specifications: www.swanspeaker.com

Model (width x height)	Imp Ω	Fs Hz	Qts	Vas Ltrs	Power Watts	dB	Frequency Range	VC Ø mm	Rec. Box Liters S / V	F3 Hz S / V	Price Each
Isodynamic Ribbon Tweeters											
RT1C-A 120mm round flange	5	-	-	-	15	94	4.5k-23kHz		Neodymium magnet		**$49.30**
RT2II 145mm x 166mm rectangular flange	8	-	-	-	30	93	2k-27kHz		Neodymium magnet		**$92.00**
RT2C-A 176mm round flange	8	-	-	-	30	93	3k-40kHz		Neodymium magnet		**$85.45**
RT2E-A 105mm x 176mm ovalish flange	8	-	-	-	30	93	3k-40kHz		Neodymium magnet		**$85.45**
RT2-Pro 220mm x 166mm rectangular flange	8	-	-	-	30	99	2.3k-35kHz		Neodymium magnet		**$180.70**
RT8II 160mm x 220mm rectangular flange	8	-	-	-	30	95	1.7k-27kHz		Neodymium magnet		**$115.00**
1" Fabric Dome Tweeters											
SD1.1-A 116mm round flange, shielded	5	1200	-	-	15	91	3k-20kHz	25	Shielded magnet		**$12.50**
Q1R 116mm round flange, shielded	6	1000	-	-	15	89	2.5k-20kHz	28	Shielded magnet		**$14.80**
TN28 46mm Ø x 99mm Top Mount	6	1300	-	-	15	90	3k-25kHz	28	Top mount, shielded		**$13.15**
TN25 54.5mm square with chamber	5	1500	-	-	15	90	3k-21kHz	25	Small flange, shielded		**$8.25**
K1 116mm round aluminum flange, shielded	5	800	-	-	15	92	2.5k-25kHz	25	Aluminum face plate		**$8.55**
X1R 116mm round flange, ferrofluid cooled	4	1600	-	-	15	91	3k-20kHz	25	decorative face plate		**$8.55**
2" Fabric Dome Midranges											
DMN-A 116mm round flange with chamber	5	800	-	-	60	93	1kHz-10kHz	50	117mm deep bullet ch.		**$26.30**
DMA-A 145mm round flange, shielded	5	630	-	-	80	92	800Hz-10kHz	50	43mm D., smooth dome		**$23.70**
DMB-A 145mm round flange, shielded	5	700	-	-	80	92	800Hz-10kHz	50	43mm D., rough dome		**$23.70**
Small Full Range Speakers, Shielded, Metal Alloy Cones											
A2S 53mm square flange, shielded, silver color	8	153	1.27	0.2	10	78	150Hz-15kHz	25	0.5 +	150	**$10.20**
B2S 53mm square flange, shielded, copper color	8	152	1.2	0.2	10	78	150Hz-15kHz	25	0.5 +	150	**$10.85**
B3S 78mm square flange, shielded, copper color	8	80	0.93	1.6	15	80	80Hz-8kHz	20	0.5 +	80	**$9.55**
B3N 90mm round flange, shielded, copper color	8	77	0.86	1.7	15	81	80Hz-8kHz	20	0.5 +	80	**$9.55**
"D" Series Mineral Filled Polypropylene Cones, Large Diameter Voice Coils											
D6.8 174mm round flange, cast frame	8	42	0.40	13.5	60	85	40Hz-4kHz	76	7 / 12	74 / 42	**$62.75**
D8.8 243mm round flange, cast frame	8	30	0.38	46.8	150	87	30Hz-2kHz	100	18 / 31	57 / 34	**$87.75**
D10.8 299mm round flange, cast frame	8	20	0.40	211.1	150	87	35Hz-500Hz	100	90 / -	35 / -	**$104.15**
"F" Series Kevlar/Paper Cones, Cast Frames, Shielded Magnets, Yellow Cones											
F5 154mm round flange, phase plug	8	52	0.33	7.3	35	86	65Hz-8kHz	25	2 / 4	111 / 63	**$36.15**
F6 197mm round flange, phase plug	8	45	0.35	16.4	45	88	50Hz-5kHz	25	5 / 11	92 / 51	**$45.65**
F8 217mm round flange	8	36	0.39	36.9	60	87	38Hz-3kHz	35	17 / 30	65 / 38	**$65.40**
"M" Series Magnesium/Aluminum Alloy Cones, Copper Color Cones, Cast Frames											
M5a 140mm round flange, shielded magnet	8	48	0.34	11.2	35	87	65Hz-3kHz	25	3 / 5	100 / 63	**$33.85**
M6a 174mm round flange, shielded magnet	8	46	0.38	13.6	45	88	50Hz-3kHz	25	9 / 14	86 / 53	**$44.35**
M8a 215mm round flange	8	30	0.43	50.3	80	87	30Hz-2kHz	35	29 / 40	49 / 32	**$58.50**
M12 313mm round flange	4	27	0.33	81.4	150	93	30Hz-1.5kHz	50	24 / 53	57 / 32	**$131.75**
Various Polymer Cone speakers											
W5 153mm round flange, shielded, phase plug	8	57	0.49	8.6	35	86	50Hz-7kHz	25	8 / 9	82 / 52	**$35.80**
W6 179mm round flange, shielded	8	40	0.52	25.1	45	86	55Hz-3kHz	25	15 / -	55 / -	**$44.70**
BG8N 222mm round flange, shielded magnet	8	37	0.36	46.1	60	89	45Hz-8kHz	35	16 / 28	72 / 45	**$43.05**
SP10 246mm round flange, subwoofer	4	34	0.57	17.1	500	84	40Hz-500Hz	76	32	40	**$170.85**

Madisound has recently become a distributor for Hi-Vi Research. We have chosen products that we feel will be most useful to the DIY market. There are many options to chose from and we are eager to learn along with you on the best applications for these drivers. Of course we will be offering Leap designs for this product, although it will take a little longer for us to measure them all.

Please let us know how we can assist you in choosing the right speaker for your project. Please visit our web site at www.madisound.com.



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e-mail: info@madisound.com
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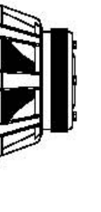

to carry direct current, it may be optimized for audio performance.

A constant-current source (CCS)—as its name implies—provides a constant direct current and very high impedance to the signal current. Additionally, a CCS provides a very high power-supply rejection ratio (PSRR). This means that the audio circuit is isolated from aberrations in the power supply.

The CCS may be a very large resistor, an audio choke, or an electronic circuit. A large resistor will drop a large voltage, requiring that the power supply have a very high voltage for the B+. An audio choke, while not producing a large DC voltage drop, is large and heavy and still does not provide the very high level of impedance that an electronic circuit provides.

A MOSFET CCS can provide an impedance in excess of 1MΩ while dropping only 10V. Such a circuit is shown in *Fig. 3*. This deceptively simple circuit is very effective. The value of the source resistor, Rset, sets the current through the MOSFET in a manner similar to a cathode bias resistor in a tube. The other resistor is a gate stopper, which inhibits any tendency for the CCS to oscillate.

DIFFERENTIAL SOLUTION

This is pretty much where I left you at the end of the SET line stage article. While the result was excellent, I knew that I could improve on it (see *Photo 1*). I had heard several push-pull line stages that sounded pretty good, so I wanted to experiment with a differential topology. Single-ended circuits, by their very nature, are unbalanced with respect to their "ground" reference.

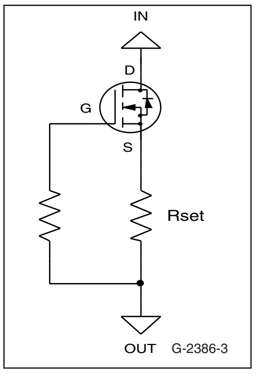

FIGURE 3: MOSFET constant-current source.

As shown in *Fig. 4*, the input signal is taken as the difference between ground and the grid, while the output is the difference between the plate and the positive terminal of the power supply. The positive terminal of the power supply is theoretically at signal (AC) ground; however, because of the limitations in physical devices, a real AC impedance exists.

To make matters worse, this AC imped-

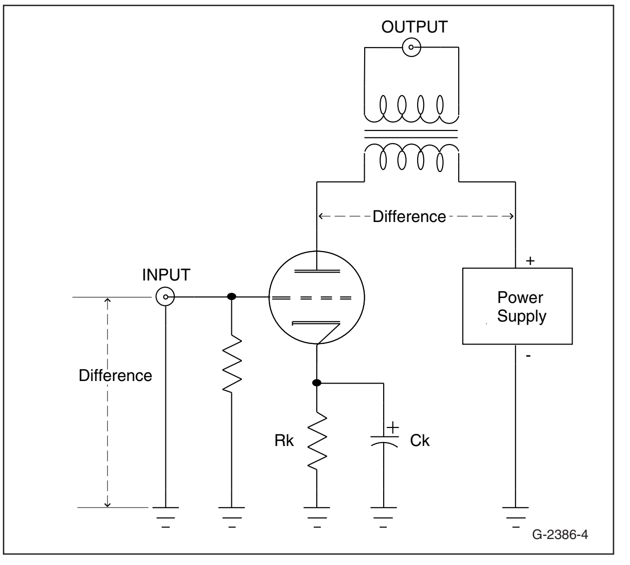

FIGURE 4: Signal references in a single-ended line stage.

ance is not fixed, but rather varies with frequency. Another problem with this circuit is that the input is susceptible to signal noise. One way to alleviate these problems is to use a differential circuit.

There are several ways to implement a differential amplifier; the classic push-pull configuration is shown in *Fig. 5*. The differential circuit is really two single-ended circuits, side by side. The important difference is the input and output references. Now the input is referenced between the two grids while the output is referenced between the two plates. This input scheme will provide better rejection of common-mode noise.

Another benefit is that distortions in the circuit, being the same on both sides, will cancel on the output. Of course, for this to work the signals on the two grids must be different—in fact opposite. An electronic circuit or a transformer may provide this inversion of the signal on one of the grids with respect to the other. Because an electronic circuit will suffer some of the ills that you are trying to remedy, I believe that a transformer is the best solution for this function. I really liked the Lundahl LL1674 that I used as a parafeed output transformer, and since this transformer was originally intended to serve as a line input transformer, it was natural for me to select it for the phase-splitting task.

PARAFEED VARIATIONS

As much as the differential circuit is an improvement over the single-ended circuit, it still suffers the earlier problem of sharing much of the circuit for the power-supply current and signal current. The obvious solution is to use a parafeed topology for the push-pull circuit (*Fig. 6*). This shows the input circuit with a transformer to provide the phase inversion. The parafeed circuit works as before with the pair of CCS, isolating the power supply from the signal and the parafeed capacitor isolating the output transformer from the power-supply current. Think of it as two single-ended parafeed amplifiers sharing a common output transformer.

In fact, I built the first prototype using a pair of the SET line-stage boards. I believed that this would give a more valid comparison when audition-

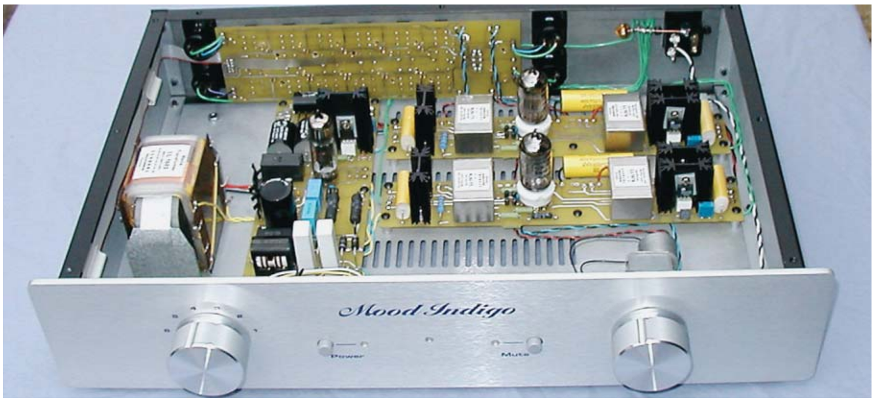

PHOTO 1: Inspired by Duke Ellington's music, which can be dynamic and driving or sweet and mellow, the author named the completed preamplifier "Mood Indigo."

ing the new line stage. The result was immediately obvious—a more lively presentation and better impact. I knew that I was on the right track.

Lynn Olson described another parafeed variation in his article "Ultrapath, Parallel feed and Western Electric" in *Vacuum Tube Valley* issue 16. I think of the circuit shown in *Fig. 7* as two single-ended-triode amplifiers, one driven in opposite phase from the other, bridged with a transformer between their plates.

I re-wired my test circuit with this circuit, but didn't expect much. After all, to my thinking, it was a minor twist on an old design. I was surprised at the audible difference between this circuit and the one I had been using earlier. I anticipated a subtle difference, but the sound was much cleaner and clearer without sacrificing any of the dynamics I had come to appreciate.

The most obvious difference between the two configurations is that *Fig. 6* has two signal current loops, while *Fig. 7* has a single signal current loop. This is significant because with the classic push-pull circuit, the transformer is mixing the two signal current loops magnetically in separate input windings. Any imbalance between the two windings or between the two signal currents will produce an aberration of the signal at the output of the transformer.

The single signal loop sidesteps this problem and may be responsible for at least some of the characteristics that SET fans applaud. Another aspect of this circuit is the advantage of a shared cathode resistor, which ensures that the two triodes are biased at the same point, further aiding circuit balance. As with the push-pull circuit, the triode is the only thing common to the power-supply current loop and signal current loop in

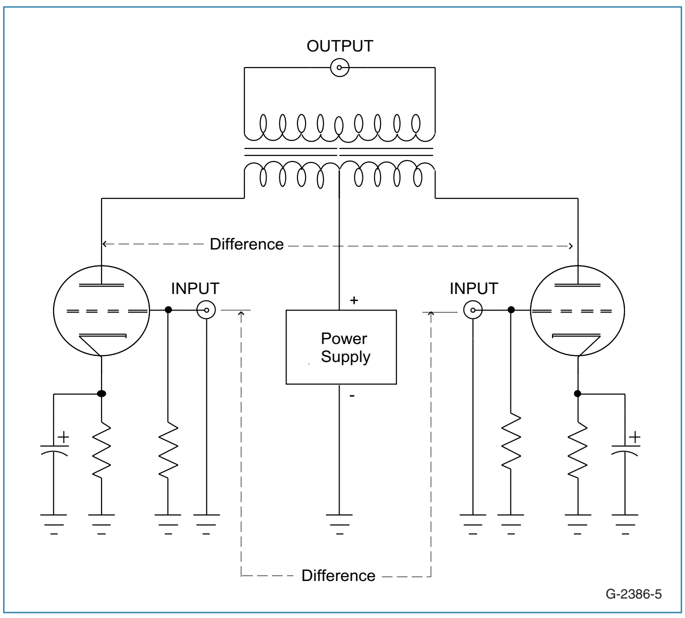

FIGURE 5: Signal references in a differential line stage.

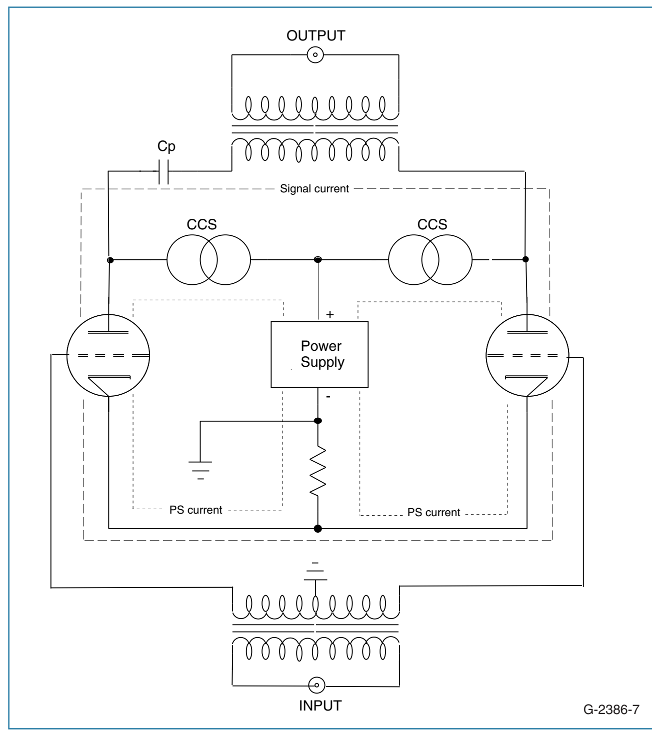

FIGURE 7: Differential parafeed line stage.

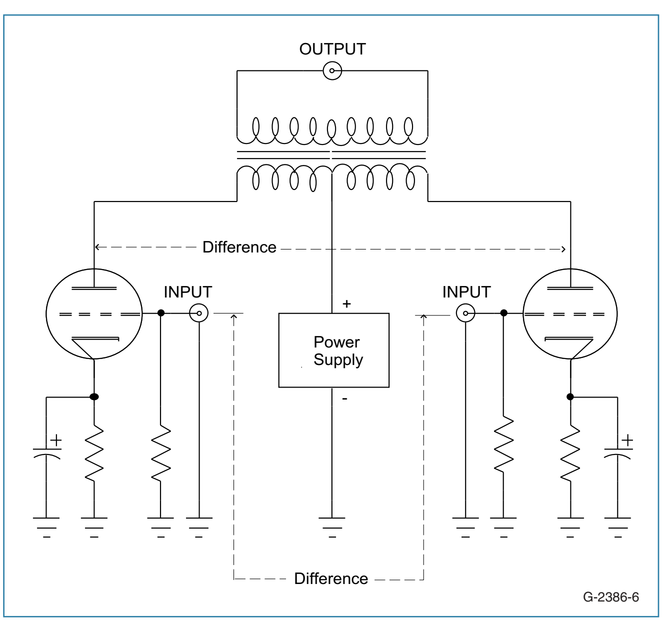

FIGURE 6: Push-pull parafeed line stage.

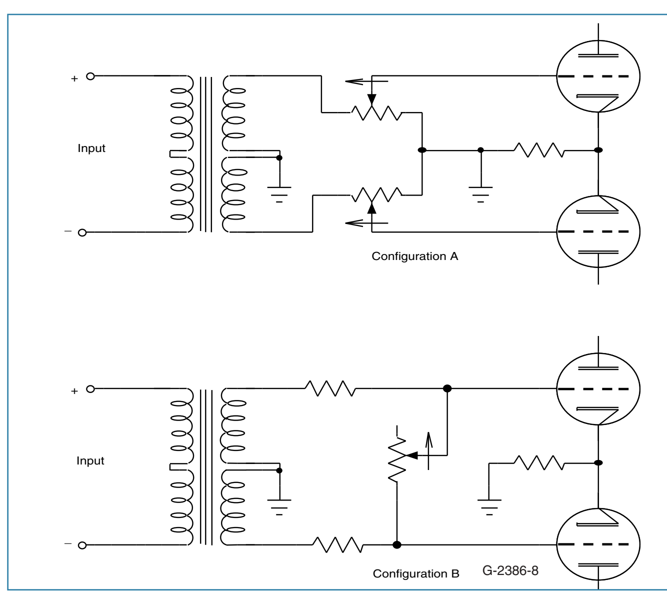

FIGURE 8: Differential volume control configurations.

the circuit shown in *Fig. 7.*

During the development of the differential circuit, I experimented with several ways of implementing a volume control, two of which are shown in *Fig. 8*. In a single-ended configuration the signal is connected to the top of the attenuator, the bottom of the attenuator is connected to ground, and the output is taken off the wiper. Configuration A shows two such attenuators, one for each phase.

TABLE 1

PARTS LIST

Part	Description	Source
Resistors		
R1, R2	5.1k 2W Kiwame	Michael Percy
R3	Note 1	
R4, R5	1k 2W Kiwame	Michael Percy
R6	121Ω Mills MRB-5	Michael Percy
R7, R9	100Ω 2W Kiwame	Michael Percy
R8, R10	200Ω Caddock MK-132	Michael Percy
R11	2.2Ω 3W	Mouser 283-2.2
R12	47Ω ¼W	Mouser 271-47
Capacitors		
C1, C3	1µF 400V GE Film	Allied Electronics 591-8015
C2	2µF Kimber Kap	Parts Connexion
C4	100nF 50V	Digi-Key P4923-ND
Inductors		
L1, L2	EMI bead	Digi-Key P9817BK-ND
Miscellaneous		
T1, T2	Transformer, Lundahl LL1674	K&K Audio
V1	Sovtek 6N1P	Triode Electronics
Q1, Q2	MOSFET DN2540 N5	K&K Audio
	LM317T	Digi-Key LM317T-ND
	PC board	K&K Audio
	9-pin tube socket	Triode Electronics
	Heatsink (qty 3)	Digi-Key HS190-ND
	Insulator, heatsink (qty 3)	Digi-Key BER102-ND
	Insulating bushing (qty 3)	Digi-Key 3049K-ND
	4-40 screw & nut (3 ea.)	Procure locally
	Mounting hardware	Procure locally

Notes

1. 10k volume control of your choice
2. A line stage kit is available from K&K Audio

PROS AND CONS

There are two advantages to this configuration. First, the impedance of the attenuator, reflected through the input transformer to the input, is constant for all settings. Second, since it takes a stereo potentiometer to implement this configuration, you can use a separate one for each channel and thereby achieve a balance function.

The disadvantage of this configuration is that it is more expensive, requiring either two stereo potentiometers or a single quad

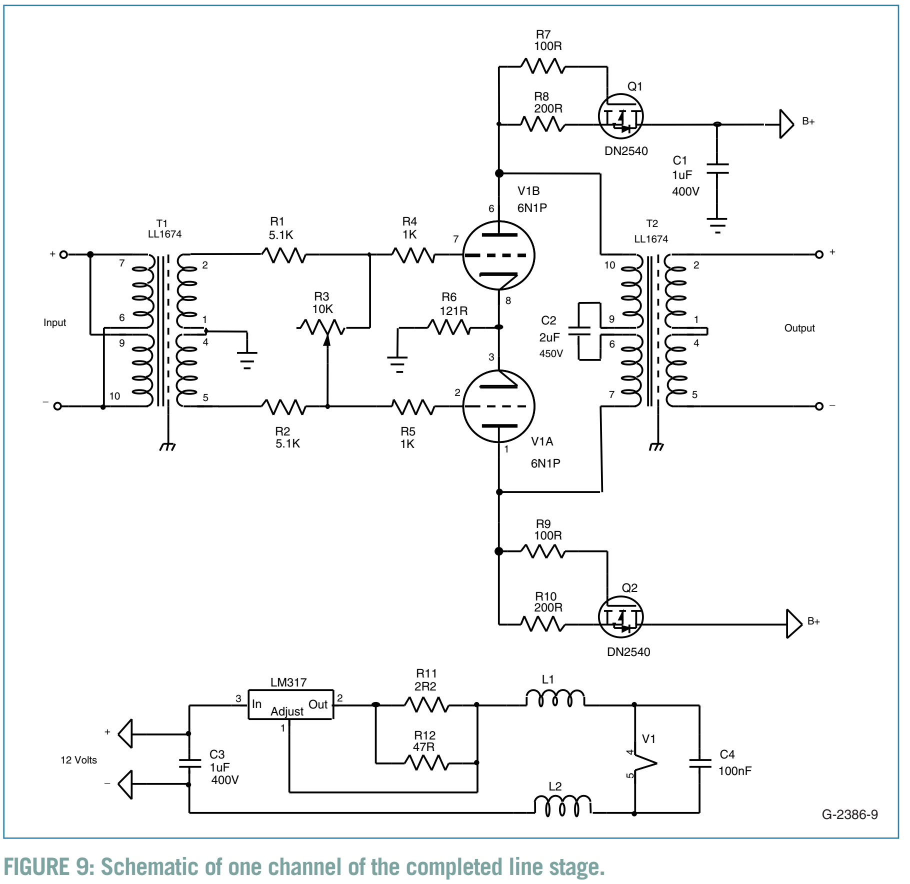

FIGURE 9: Schematic of one channel of the completed line stage.

potentiometer to implement the volume control function. The attenuator shown in configuration B takes a different approach. Decreasing the resistance in the potentiometer attenuates the differential signal on the two grids. This decreasing resistance is reflected back through the input transformer so that, unfortunately, the input impedance varies with the setting of the potentiometer. However, the two resistors located between the transformer and potentiometer limit the minimum impedance to an acceptable value.

The advantage of this configuration is that it requires only a single stereo potentiometer to implement the volume control functions for both channels. Aside from the technical differences, there are audible differences between the two configurations. I found that I prefer the sonic characteristics of configuration B over those of configuration A as well as those of several others I tried. Perhaps—and I am speculating here—there is something going on here similar to that in the output with one signal loop versus two signal loops.

There are two signal loops in configuration A—one for each half of the differential circuit. Each loop goes from ground, through one-half of the transformer, through a potentiometer, and back to ground. Differences in the two halves of the transformer as well as between the two potentiometers will affect the differential balance. There is a single signal loop in configuration B, through the transformer, through the two resistors, and through the potentiometer. Since everything is in series, differences between the two halves of the transformer or between the two resistors do not produce an unbalance in the differential signal.

HEARING IS BELIEVING

During the development of the SET line stage, I experimented with adding a small amount of load across the secondary of the output transformer to quiet a small amount of very high frequency ringing in the transformer. This use of a load is controversial: some swear by it, while others condemn it. I must say that in cases like this I let my ears be the final arbiter. To me, a little load sounded better, so I used it and carried the load into the differential design with a resistor across the secondaries of both the input transformer and the output transformer.

Early on in the development of the differential line stage, it also sounded better with the resistors present. However, as the design evolved through many refinements, it progressed to the point where it sounded better without those resistors. So, the lesson learned here is that some designs may benefit from adding a secondary load; however, I believe that it is really compensating or masking problems elsewhere in the circuit.

I had been using the 6H30 dual triode, which I selected after listening to many candidates during the SET line stage development. Kevin Carter of K&K Audio, the US Lundahl distributor, told me that he had heard good things about the 6N1P dual triode. Looking at the data sheets, I thought, "no way can the 6N1P outperform the 6H30, it doesn't have the muscle." However, I learned long ago not to listen to the data sheets, so I decided to try it anyway.

It was a simple enough experiment—the pin configuration was the same, and a switched resistor was all that I needed to change the idle current. Well, the 6N1P may not have the muscle of a 6H30, but it has the finesse. The music was clearer and more detailed. There was no indication that it was deficient in the power category—the dynamics were excellent. The difference was subtle. I still think the 6H30 is an excellent performer, which I wouldn't hesitate to use where I needed its drive capability, but for this application the 6N1P is the clear winner.

Note the schematic diagram in *Fig. 9*. Most of what you see there will now be familiar, although there are a couple of minor points not yet considered.

As you remember from the SET article, I was plagued by problems of oscillations in the circuit and went to great lengths to eliminate them. In addition to gate stoppers and grid stoppers, I added plate stoppers and drain stoppers. I am not sure why, but the differential circuit is much better behaved in this respect, and only gate stoppers (R7, R9) and grid stoppers (R4, R5) are required to curb any tendency to oscillate.

Lastly, the circuit at the bottom of the schematic diagram provides a regulated filament current for the tube. *Photo 2* shows this circuit packaged on a printed circuit board, the layout for which is shown in *Figs. 10* and *11*.

FINE-TUNING

Having chosen the circuit configuration and tube, which sounded really good, I still had a few things that I wanted to experiment with before I put the lid on the project. The differential line stage was now even more revealing and sonically clearer than the excellent SET version I had been using. I could easily hear a difference between various passive components, so it was time to reevaluate the components I was using.

The capacitors were easy to choose: C1, C3, and C4 are there for noise decoupling and have little, if any, effect on the sonics of the line stage. The parafeed capacitor, C2, definitely does have an effect and must be a high-quality capacitor. My preference here is the Kimber Kap; however, your ears are different from mine, so you may prefer the characteristics of another brand.

Choosing the best-sounding resistors was an entirely different matter. It took many hours of critical listening to audition various combinations of different types of resistors in the nine locations other than the potentiometer and those in the filament regulator circuit.

My first thought was, "but where to start?" Well, tantalum resistors have a reputation for being the best, so that is where I chose to begin this leg of the journey. I populated all locations with tantalum resistors and let the line stage

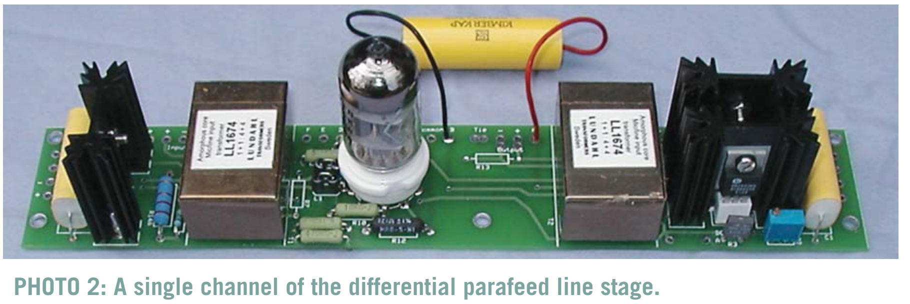

PHOTO 2: A single channel of the differential parafeed line stage.

play for a few hours to let everything settle in. It sounded great! The word "aristocratic" came to mind to describe the sound of these resistors.

As good as it sounded, I was not about to let well enough alone—I replaced all of the resistors several times, first with carbon film, then exotic metal, then wire wound, and several others. None sounded as good as the tantalums. Was I done? No, not quite.

Working from a base of tantalum resistors, I replaced each resistor one at a time with each of the different types. So, for example, I would have eight tantalum resistors and one carbon film resistor. This stage of the experiment proved very interesting and worthwhile. It turns out that different locations are best served by different kinds of resistors.

When I was finished testing, I had a mix of Caddock, Kiwame, and Mills resistors with no tantalum resistor in sight. My final resistor choices are included in *Table 1*.

For months I had the line stage mounted vertically on the outside of a

SOURCES

Allied Electronics
7410 Pebble Dr.
Fort Worth, TX 76118
800-433-5700
www.alliedelec.com

Digi-Key
PO Box 677
701 Brooks Ave. S.
Thief River Falls, MN 56701 0677
218-681-3380
FAX 218-681-3380

K&K Audio
1004 Olive Chapel Rd.
Apex, NC 27502
919-387-0911

Michael Percy Audio Products
11731 Stillwater Creek Rd.
Nevada City, CA 95959
530-470-8650
FAX 530-470-8651

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

Parts Connexion
2885 Sherwood Heights Dr.
Oakville ON L6J 7H1
Canada
905-681-9602

Triode Electronics
5633 W Irving Park Rd.
Chicago, IL 60634
773-871-745

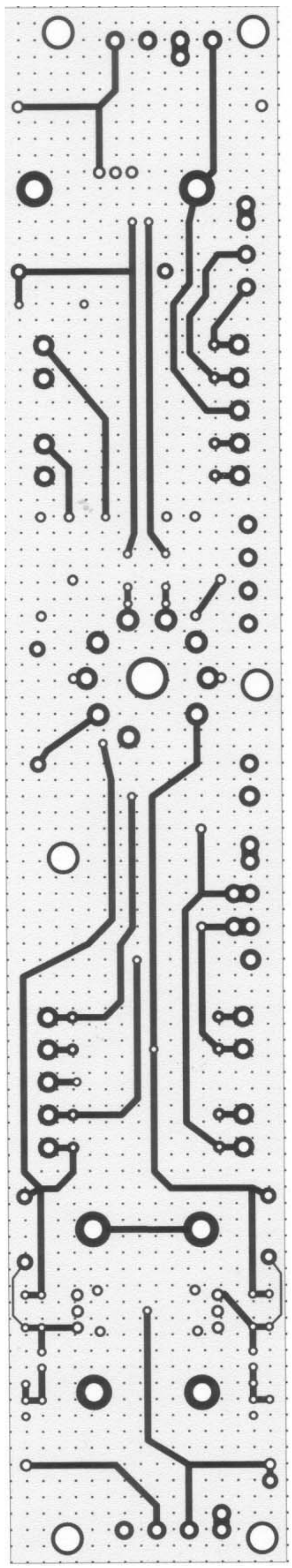

FIGURE 10: Top layer of the printed circuit board.

metal box to facilitate changes and experiments. But deciding to offer a full preamplifier kit (which is a story for another time) forced me to do things properly—not cutting any corners—and I ended up with a full-function, professionally finished high-end preamplifier for my own use. The result, which is shown in *Photo 3*, in addition to a pair of line stage boards and minimal reactance power supply, has remote-controlled volume control and mute as well as a mix of relay-switched RCA and XLR connectors for input and output.

The conclusions drawn at the end of the second part of the SET line stage article are for the most part still valid. In addition, I found that the 6N1P is an excellent candidate for a line stage. Most significantly, I found the differential configuration sounded better than the original SET version of the line stage, and furthermore, the bridged SET is the premier differential configuration.

The odyssey has been a lot longer than I ever imagined when I started, taking me to new areas, allowing me to learn many things and providing much enjoyment along the way. I hope that you, too, have enjoyed the journey and perhaps picked up a few ideas for your next project. Bon voyage! ❖

PHOTO 3: Another view of the preamplifier showing the input and output arrangement.

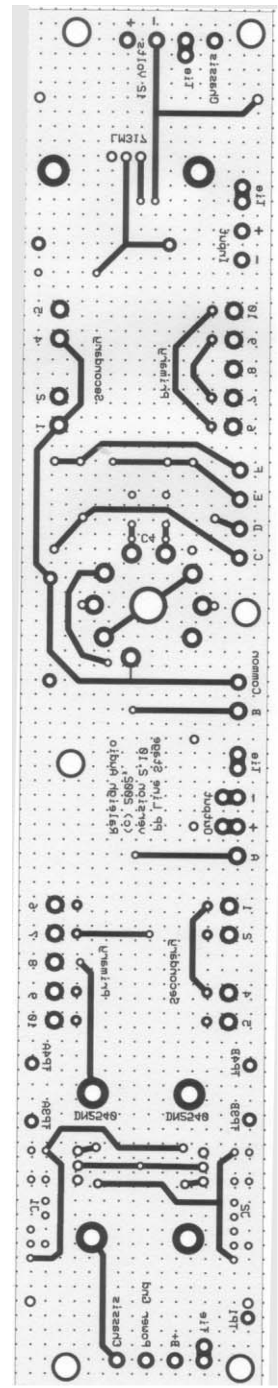

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FIGURE 11: Bottom layer of the printed circuit board.