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The cabinets are made of 5/8" MDF finished in an unobtrusive "black ash" vinyl laminate and include grills with black cloth. All driver holes are precision cut with a CNC for a perfect fit. The tweeter is flush mounted to reduce diffraction effects.

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Audio News

FROM FOCAL PRESS

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SOUNDELUX E47

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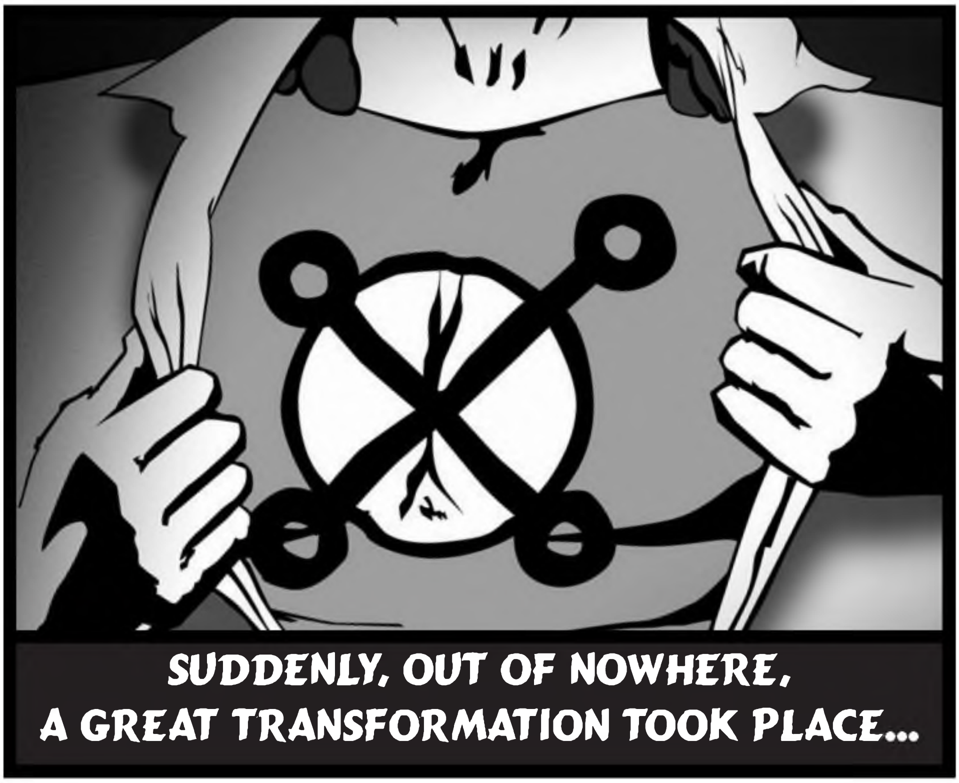
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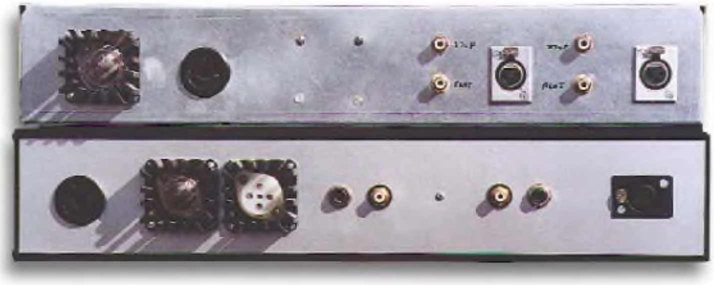

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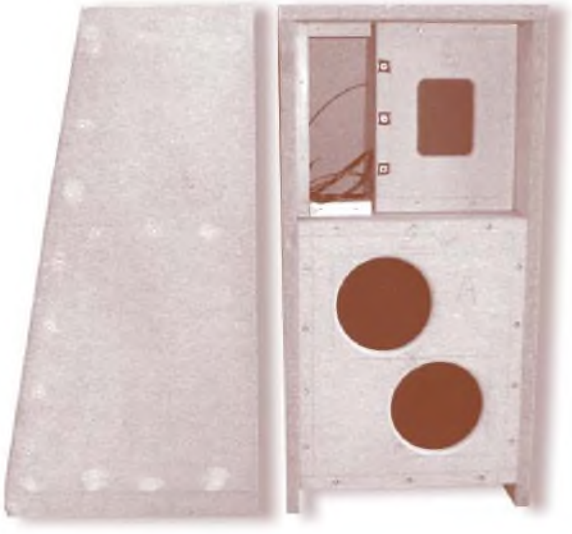

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> *The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it.*
>
> JOHN STUART MILL

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Big Mike and the Jimmy, Part 1

PHOTO 1: Front view of mike preamp.

Back by popular demand, and 20 years in the making, meet Big Mike, a tube-based microphone preamp design. **By Paul J. Stamler**

A few years ago, I designed a solid-state microphone preamp. It sounded good, and I wrote about it for *Audio Amateur*[1] and, later, *Recording*[2]. In the process, I incautiously let slip that I was also working on a tubed design.

Tubes are hot these days (well, they would be, wouldn't they?), and readers immediately asked for details. So I plugged away at the design, which was as good a preamp as I knew how to produce. It's taken a while—in fact, 20 years, all told. Now it's done; I call it Big Mike (*Photo 1*).

WHAT DOES A MIKE PREAMP DO?

As usual, I laid out a bill of particulars for the design. What would I expect it to accomplish, and under what conditions?

- The mike preamp should amplify microphone signals of every level I would expect to encounter to line level, either the consumer standard (−10dBV) or pro-level (+4dBu), without significant or audible distortion.
- It should not add significant noise to the microphone's inherent, self-generated noise.
- The preamp should provide "phantom power" to those mikes that need it for powering internal amplifiers.
- There should be a switchable high-pass filter to remove room rumble and/or to compensate for proximity effect when microphones are used close up.
- The preamp should be unaffected by radio-frequency interference from the outside world, and by garbage carried on the AC power line.

I decided that would be it; I wouldn't incorporate extraneous features such as EQ controls or polarity reversal. You can always add EQ downstream—usually more effectively—with an outboard box, and on the rare occasions when I need polarity reversal, a short cable with the wires switched does perfectly well without adding switch contacts to the normal signal path.

In fact, I decided to keep the signal path as clean and simple as possible, in keeping with standard audiophile practice. It would include just two active stages, a level control, and two switches (one for the high-pass filter, one for phantom power). Period.

I would also regulate all power supplies as tightly as possible—for the sake of sonics, but also to isolate the signal circuits from line garbage and surges. I'd mount the power transformers in an outboard box to minimize inductive hum pickup. And I would try to isolate the tubes from vibration to minimize microphonic pickup.

I wound up with a design conceptually similar to the solid-state preamp (*Photo 2*): two fixed-gain amplifiers with a level control between them. In tube preamps this is a bit unusual; it's more common to use a variable-feedback circuit in the input stage to control gain. Here's why I didn't do that.

PHOTO 2: Rear panel.

INPUT CONSTRAINTS

Begin with a typical dynamic or ribbon microphone, with output low enough that preamp noise becomes an issue. Typical mikes of this sort present a source impedance to the preamp of 150Ω, and this impedance, all by itself, generates about 0.22µV of noise, or about −130.8dBu. (All noise figures assume a signal bandwidth of 20kHz.)

With tubes, you need a transformer at the input, so assume for the moment that it's a Jensen JT-115K-E, a device I've used before and know well. The turns ratio is 1:10, so the impedance presented to the input tube will be 15k, with a corresponding noise level of 2.2µV.

Ideally, you want to add as little noise as possible. In practice, let's set an acceptable noise level of 1dB, over and above the microphone's inherent noise.

Bipolar junction transistors possess two noise mechanisms: voltage noise and current noise. Field-effect transistors and triodes are simpler—to all intents and purposes, they have only volt-

ABOUT THE AUTHOR

Paul J. Stamler is a recording engineer/producer, musician, and technical writer; he also hosts a radio program, "No Time to Tarry Here," featuring traditional folk music and related stuff. He has delighted in 78s since he was a boy, when they were still being made.

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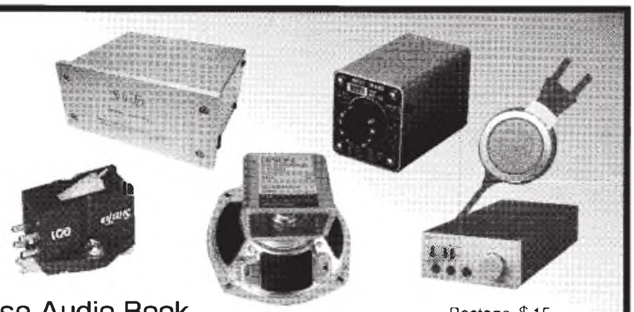

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■MC STEP UP TRANS

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

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■Speaker

Model	Specifications					Price*	Postage** (US$)			
	D(cm)	Ω	Response	db	w	(US$)	I	II	III	IV
Diatone P-610MB	16	8	45Hz～20kHz	90	7	360	30	40	50	66
Fostex FE208Σ	20	8	45Hz～20kHz	96.5	100	296	62	74	120	156
Fostex FE168Σ	16	8	60Hz～20kHz	94	80	236	42	50	73	98
Onken OS5000T	—	8	7kHz～25kHz	105	2.5	4,000	70	84	133	181
ALE 1710 Tweeter	8	16	6kHz～	118	10	3,380	85	110	170	230

*Price is for a pair **Air Economy

■STAX

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

■TANGO TRANS(28 models are available now)

Model	Specifications				Price (US$)	Postage** (US$)			
	W	Pri.Imp(kΩ)	Freq Response	Application		I	II	III	IV
XE-20S(SE OPT)	20	2.5, 3.5, 5	20Hz～90kHz	300B,50,2A3	396	47	56	84	113
U-808(SE OPT)	25	2, 2.5, 3.5, 5	20Hz～65kHz	6L6,50,2A3	242	42	50	73	98
XE-60-5(PP OPT)	60	5	4Hz～80kHz	300B,KT-88,EL34	620	62	74	115	156
FX-40-5(PP OPT)	40	5	4Hz～80kHz	2A3,EL34,6L6	320	47	56	84	113
FC-30-3.5S(SE OPT) 〔XE-60-3.5S〕	30	3.5	20Hz～100kHz	300B,50,PX-25	620	62	74	115	156
FC-30-10S(SE OPT) 〔XE-60-10SNF〕	30	1.0	30Hz～50kHz	211,845	620	62	74	115	156
NC-14(Interstage)	—	[1＋1：1＋1] 5	25Hz～40kHz	[30mA] 6V6(T)	264	30	40	50	70
NC-16(Interstage)	—	[1＋1：2＋2] 7	25Hz～20kHz	[15mA] 6SN7	264	30	40	50	70

Price is for a Pair

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■TAMURA TRANS(All models are available)

Model	W	Pri.Imp(kΩ)	Freq Response	Application	Price (US$)	I	II	III	IV
F-7002(Permalloy)	10	3.5	15Hz～50kHz	300B,50	740	60	70	110	145
F-7003(Permalloy)	10	5	15Hz～50kHz	300B,50	760	60	70	110	145
F-2013	40	10	20Hz～50kHz	211,242	730	70	84	133	181
F-5002(Amorphous)	8	3	10Hz～100kHz	300B,2A3	1276	65	80	120	160

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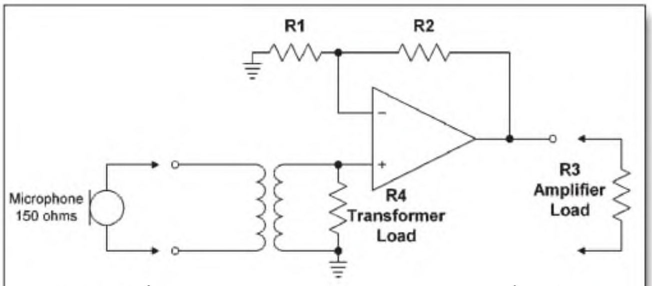

FIGURE 1: Microphone preamp—conceptual drawing. In practice, the amplifier could be differential, as shown, or single-ended, with R1 representing the first stage's cathode resistor. R2 might be variable, for gain control.

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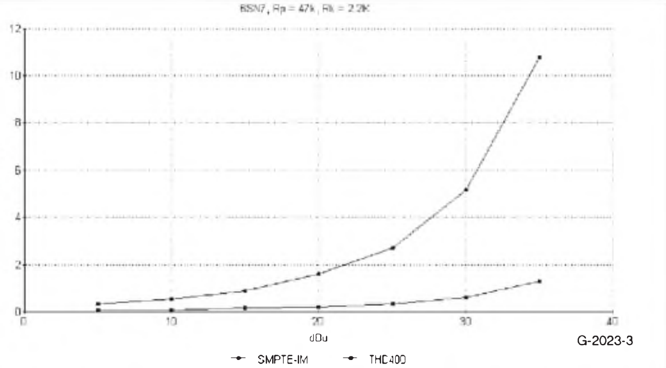

FIGURE 3: Distortion in voltage amplifier, driving a 50k load.

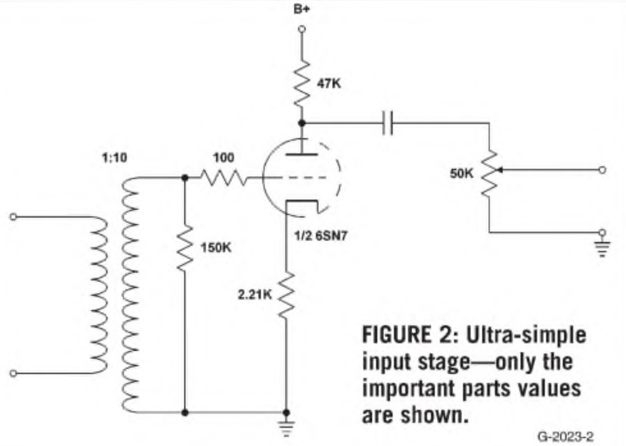

FIGURE 2: Ultra-simple input stage—only the important parts values are shown.

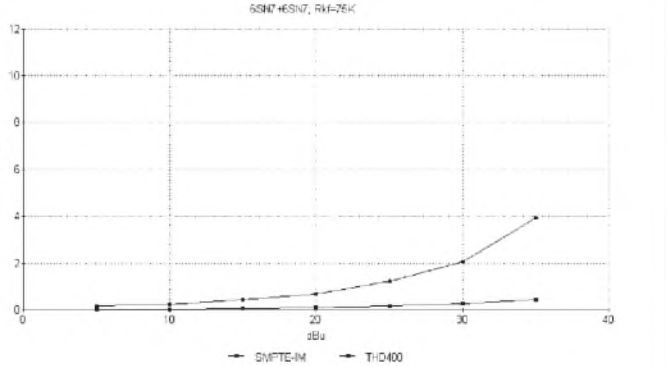

FIGURE 4: Distortion with cathode follower, driving a 50k load.

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age noise. The input noise can therefore be modelled as a "noise resistance"—that resistance which, when measured in a 20kHz bandwidth, produces the same noise as the tube.

Noise resistance makes the design process much simpler; in effect, you simply add up the resistances to find the total noise. A triode circuit typically has four noise sources:

1) Noise generated by the source resistance
2) Noise generated by the tube itself
3) Noise generated by the tube's plate resistor
4) Noise generated by the resistance on the input tube's cathode

If you want to add only 1dB of noise to the inherent noise of the microphone, you may increase the total noise resistance by only about 25.9%. That constrains the design process.

THE FEEDBACK MODEL

Take a look at *Fig. 1*, a conceptual drawing of a feedback amplifier. It could be a differential-input amp, like IC op-amps, or it could be a Dynaco PAS-type feedback pair (R1 would then represent the input tube's cathode resistor). Assuming adequate open-loop gain, the closed-loop gain will be about (R2/R1) + 1, the resistance presented to the inverting terminal will be R2 || R1 (the parallel combination of R2 and R1), and the load presented to the amplifier will be (R1+R2) || R3, where R3 is the volume control or input resistance of the next stage.

For the moment, assume you have a perfect transformer. (Not true, but leave it for now.) Also assume the amplifier clips at about +34dBu. (That's quite generous, even given a 330V B+ supply, but let that pass, too.)

I said before that a 1:10 transformer ideally presents a source impedance of 15k to the input tube. That's not actually correct; the transformer is terminated by a 150k resistor (R4), so the tube sees an effective source resistance of about 13.6k. To keep the added noise below 1dB, you can add only 25.9% to that, or about 3530Ω.

Let the input tube be a 12AX7, operating at 0.7mA with a 150k plate resistor (the Dyna operating point). According to my *Audio Designer's Tube Register*[3] data, the transconductance of the tube (g_m) will be about 1.5ms. The equivalent noise resistance of a triode is, roughly, $2.5/g_m$, or (in this case) about 1.67k.

The effective resistance on the plate is 150k in parallel with the next stage's input resistor, 1.2M, or about 133k. To get the equivalent noise resistance, divide this by the square of the tube's open-loop gain, or (mumble, mumble) about 900, resulting in a noise resistance of about 148Ω.

What's the total so far? Roughly 1815Ω. That leaves 1715 to play with, and you'll use them up in the feedback circuit. Let the closed-loop gain be 10× for the moment; now R2 = R1 × 9, so the parallel combination will be 0.9 × R1. If you want that to total 1715Ω, then R1 will be about 1900Ω and R2 about 17.1k. The total load on the last stage of the input circuit, assuming a 50k level control follows it, is about 19k || 50k, or about 13.8k.

That's a pretty steep load for a tubed small-signal amplifier. Assuming a clipping level of +34dBu out, that's 54.9V peak, and just under 4mA of current. A feedback pair or differential-input amp would need a pretty hefty output stage to do that.

Can you lower the minimum gain? No, because that makes R2 smaller, making the load worse. All right, then, can you raise the gain? Not realistically. In fact, 20dB is already high.

Russell Hamm[4] measured the output of microphones in typical studio applications and discovered surprisingly high levels. For example, a Neumann U-87 (the hottest mike he tested) produced 0dBu when it was placed 6″ from a loud yell. A mike preamp with a voltage gain of 40dB (20dB from the transformer, 20dB from the electronics) would need to put out +40dBu to avoid clipping at that level—and new microphones are hotter than the U-87.

CONCEPTUAL QUALMS

I have another problem with variable-feedback amplifiers: they are inconstant. Changing the stage gain from 20dB to 50dB changes everything: distortion characteristics, bandwidth, out-

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ECC88	5.70	EL509/519	13.00	6336A	48.00	Ditto, Gold Pl.	3.00
ECF82	5.50	E84L/7189	7.50	6550WA/WB	15.00	Octal (Ch or PCB)	1.80
ECL82	6.00	KT66	11.00	7581A	12.00	Ditto, Gold Pl.	4.20
ECL86	6.30	KT66R	22.50	807	10.70	UX4 (4-Pin)	3.60
EF86	6.00	KT77	13.20	811A	11.80	Ditto, Gold Pl.	5.50
E80F Gold Pin	11.00	KT88	13.50	812A	31.00	4 Pin Jumbo	10.00
E81CC Gold	8.00	KT88 (Special)	17.00	845 (New des)	33.50	Ditto, Gold Pl.	13.00
E82CC Gold	9.00	KT88 (GL Type)	30.00	RECTIFIERS		5 Pin (For 807)	3.30
E83CC Gold	8.50	PL509/519	9.90	EZ80	5.10	7 Pin (For 6C33C)	4.70
E88CC Gold	8.80	2A3 (4 pin)	15.50	EZ81	6.00	9 Pin (For EL509)	5.00
6EU7	7.00	2A3 (8 Pin)	17.50	GZ32	15.50	Screen can B9A	2.20
6SL7GT	8.90	211	23.00	GZ33	15.50	Ditto, Gold Pl.	4.30
6SN7GT	5.30	300B	45.00	GZ34	7.20	Top Con. (For 807)	1.70
6922	6.40	6C33C-B	25.00	GZ37	15.50	Ditto, (For EL509)	2.00
7025	7.00	6L6GC	7.60	5U4G	6.30	Retainer (For 5881)	2.20
		6L6WGC/5881	8.90	5V4GT	5.00		

******** And a few 'Other Brands', inc. rare types ********

5R4GY *Fivre/GE*	8.50	6SL7GT *STC*	13.00	13E1 *STC*	100.00	6550C *Svetlana*	18.00
5R4WGY *Chatham*	10.50	6SN7GT *Brimar*	13.00	211/VT4C *GE*	120.00	6146B *GE*	18.50
5Y3WGT *Sylv.*	6.50	12AT7WA *Mullard*	6.00	300B *JJ*	56.00	A2900 *GEC*	15.00
6AS7GT *Sylv.*	12.00	12AU7 *Mullard*	12.50	300B *Svetlana*	80.00	E88CC *Mullard*	14.60
6AU6WC *Sylv.*	5.10	12AY7 *GE / RCA*	8.40	300B *WE*	195.00	F2a *Siemens*	145.00
6B4G *Sylv.*	27.00	12AZ7 *West'h.*	8.00	805 *USA*	52.00	KT66 *GEC*	69.00
6BW6 *Brimar*	5.40	12BH7A *RCA*	14.00	5842A *GEC*	15.00	KT88 *JJ*	17.40
6BX7GT *GE / RCA*	9.00	12BY7A *GE*	9.50	6080 *Telef.*	13.30	KT88 *Svetlana*	35.00
6CG7/6FQ7	8.50	12E1 *STC*	12.50	6550A *GE*	31.50	PX25 *KR*	128.00

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put impedance. Even the degree to which capacitor nonlinearities are reduced changes. In effect, you have an infinite number of preamps, depending on the sensitivity of your microphone.

I don't like that approach. I want a preamp that behaves about the same with a hot condenser mike as with a low-output dynamic.

I decided, therefore, to make the mike preamp from fixed-gain stages, with no global feedback. The input stage would be followed by a level control, then another gain stage, and that would be it. Of course, the stages would need to be pretty clean to operate without feedback.

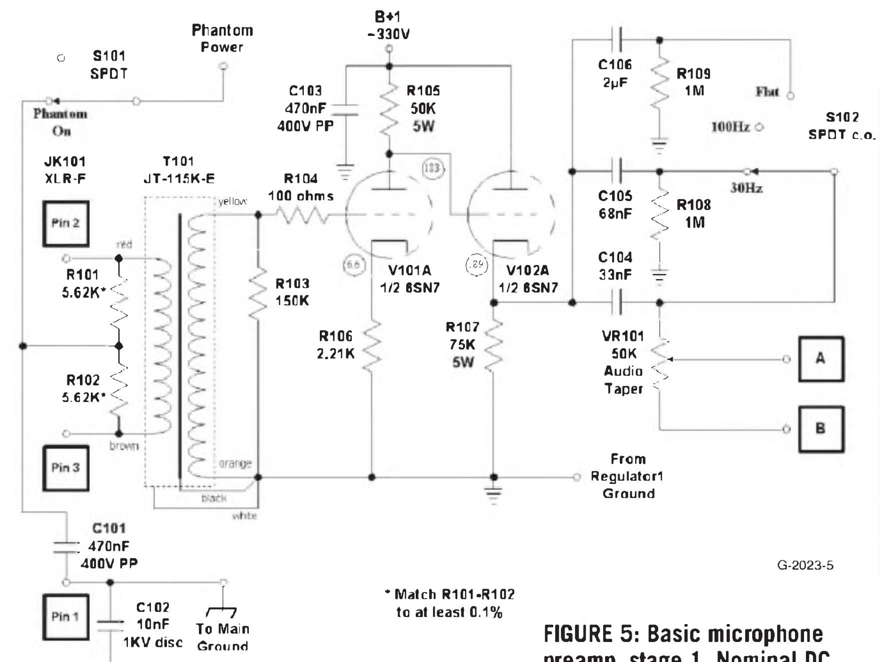

FIGURE 5: Basic microphone preamp, stage 1. Nominal DC voltages are circled.

NICE TRY, NO CIGAR

My first thought was to make the input stage from a single tube, Zen style. I needed a tube that was extremely linear, low gain (to keep the input clipping point as high as possible), and low noise. After some lab work, I decided I'd found it in the venerable 6SN7, and designed an input stage (*Fig. 2*).

I ran THD tests on this stage, and it came out well—apparently clean, good clipping point, and the harmonics were all low-order, mostly second. Then I ran SMPTE-IM tests (*Fig. 3*), and it was back to the drawing board. The SMPTE-IM test reveals buried bodies that THD tests don't touch.

OLD RELIABLE

Zen having failed me, I reverted to a configuration that has worked well for me in the past: a voltage amplifier direct-coupled to a cathode follower. This has several advantages. The high input impedance of the cathode follower provides minimal loading to the voltage amplifier; in effect, its only load is the plate resistor. This minimizes the current the voltage amplifier must supply.

The cathode follower also preserves bandwidth. Its input has low capacitance, about 4pF for a 6SN7, and the follower's lack of gain means there's no Miller effect multiplying that capacitance. The follower itself has an output impedance of about 600Ω, which is low enough that stray capacitances won't cause rolloff. (And, looking ahead, that low impedance in an output circuit will drive reasonable cable lengths without losing highs.)

Would the distortion be low enough? Indeed it would (*Fig. 4*). The THD is lower, and the SMPTE-IM is too. Indeed, I suspect there's some distortion cancellation going on between the voltage amplifier and the cathode follower.

So I had my gain block.

FRONT END

Figure 5 shows the input stage. It's quite straightforward; R101 and R102 couple phantom power to the input lines, and R104 supplies a bit of additional RFI-prevention. The cathode of the cathode follower sits a couple hundred volts above ground, so its filament needs to be hoisted to a similar level; I'll talk about that when I cover the power supplies.

The transformer, as mentioned, is a Jensen JT-115K-E. I've used this device in several designs, with excellent results. The late Deane Jensen designed it with a Bessel characteristic in the high frequencies, which translates to clean, smooth, overshoot-free transient response.

At the output of this stage, there's a switch to select capacitors for the bass rolloff. With the switch open (center-position), the 33nF coupling capacitor rolls the bass down 6dB/octave at 100Hz, a good point for combating proximity effect. Switching in the parallel 68nF cap lowers the rolloff frequency to 30Hz, good for filtering out subsonic crap without disturbing the music too much, while switching in 2μF extends bass response to 1.6Hz, effectively flat.

The level control is critical; using a cheap pot here will guarantee grief in the future if it decides to produce a dirty spot in the midst of a perfect take. Besides, low-grade pots sound crummy; they are, in effect, carbon-comp resistors with sliders attached. At a minimum, I recommend a Bourns conductive-plastic pot, and if you can afford a high-grade Alps or Noble unit, by all means use one. (A stereo 100k pot, with the two sections hooked up in parallel, gives excellent results.)

SECOND STAGE

Stage 2 (*Fig. 6*) is essentially the same as stage 1, without the switched capacitors. Notice the changed value of the cathode-follower's resistor; this provides cleaner performance into lower impedances, allowing the stage to drive the 10k load typical of an ADAT input. (I'll talk about that in Part 3.)

Next month I'll look at the power supplies. ❖

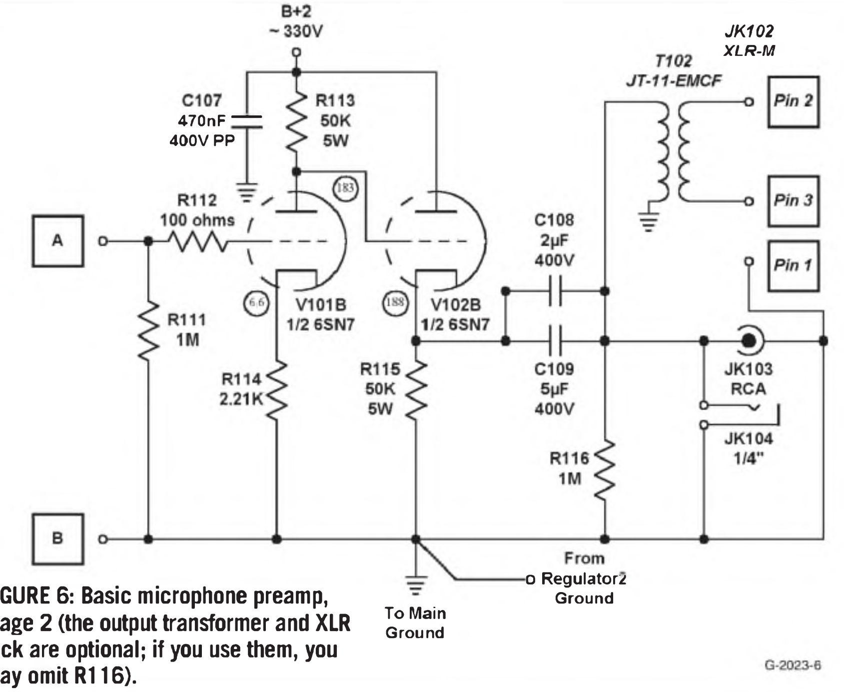

FIGURE 6: Basic microphone preamp, stage 2 (the output transformer and XLR jack are optional; if you use them, you may omit R116).

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3. Tom Mitchell, *The Audio Designer's Tube Register, Vol. 1,* Media Concepts, Norwalk, CA 1993.
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Panel Damping Studies: Reducing Loudspeaker Enclosure Vibrations

Here's an extensive study to try to answer the age-old question: "How best to build a box to minimize vibrations?" **By Jim Moriyasu**

The design and construction of an enclosure is one of the most challenging tasks facing loudspeaker builders. What is the best way to construct an enclosure that is visually appealing yet acoustically inert? In the past, I have built enclosures out of particleboard, plywood, and medium density fiberboard (MDF). I've installed the odd brace or two and have applied various damping strategies to minimize the resonance of the open panels.

I used to think 1/16″ lead sheeting glued to a panel was the ultimate panel-damping treatment since it produced (apparently) favorable results when I rapped it with my knuckles. Still, I did not really know what design techniques, materials, or treatments were the most effective. However, the acquisition of an AMP ACH-01 accelerometer coupled with the impulse measurement capability of Liberty Audiosuite (Laud) and a good dose of curiosity led to this odyssey on panel vibrations.

RESONANCE MODES

According to theoretical analysis, panels can have multiple modes of vibration or resonance.[1] The primary mode is characterized as an in-and-out movement of the panel. The second mode can be visualized by dividing the panel in two vertically, for example, and imagining half the panel moves away while the other half moves forward. The third mode divides the panel in two horizontally.

The fourth mode divides the panel into thirds, and here the middle section moves forward as the other sections move backward. The fifth mode divides the panel into fourths and is characterized by two diagonally opposing quarters moving forward as the other two move backward. Higher modes are carried out by further subdivision of the panel.

TEST SETUP

In order to isolate the effect of various damping methods, I built a test box with a removable 14[illegible]/16″ panel, which is held in place during testing with eight toggle clamps (*Photo 1*). The panel is supported by a frame of 3/4″ thick, one-inch-wide MDF. Thus, there is approximately a 12″ × 12″ square panel that is subject to vibration. A Peerless 831858 8″ woofer provides the excitation; it is vented to provide a low-frequency cutoff (f_3) of approximately 25Hz.

The enclosure has a volume of 1.4ft^3 and is built from 3/4″ MDF with no internal bracing except for the frame that holds the test panel. The woofer is secured to the front baffle with 10/32 × 1″ hex-socket screws threaded into brass threaded inserts. I inserted a one-piece 1/32″ cork/neoprene gasket between the woofer and the front baffle to ensure an airtight seal, and also attached a one-piece 1/32″ neoprene sheet rubber gasket to the test cabinet to ensure an airtight seal for the test panel. I vertically positioned the enclosure on three spikes and placed it on a concrete floor.

I glued the accelerometer to a 1¾″ × 1⅛″ × ½″ acrylic block with cyanoacrylate glue and then affixed the block to the middle of the test panel with 1½″ wide double-sided general purpose carpet tape. I used a fresh piece of tape for each test, since it loses some effectiveness once it is removed (*Photo 2*).

I set the output of the Laud to 3.36mV RMS/√Hz, ~1.101Vpk and fed it to the NAD 2140 power amplifier that multiplies voltage by 15.35 times. I set the main in-level to 22.50dB, and set the window as wide as possible, 84.7ms, with a sample size of 16384 points and a sample rate of 48.0k.

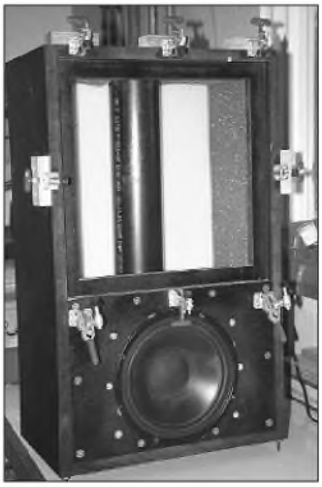
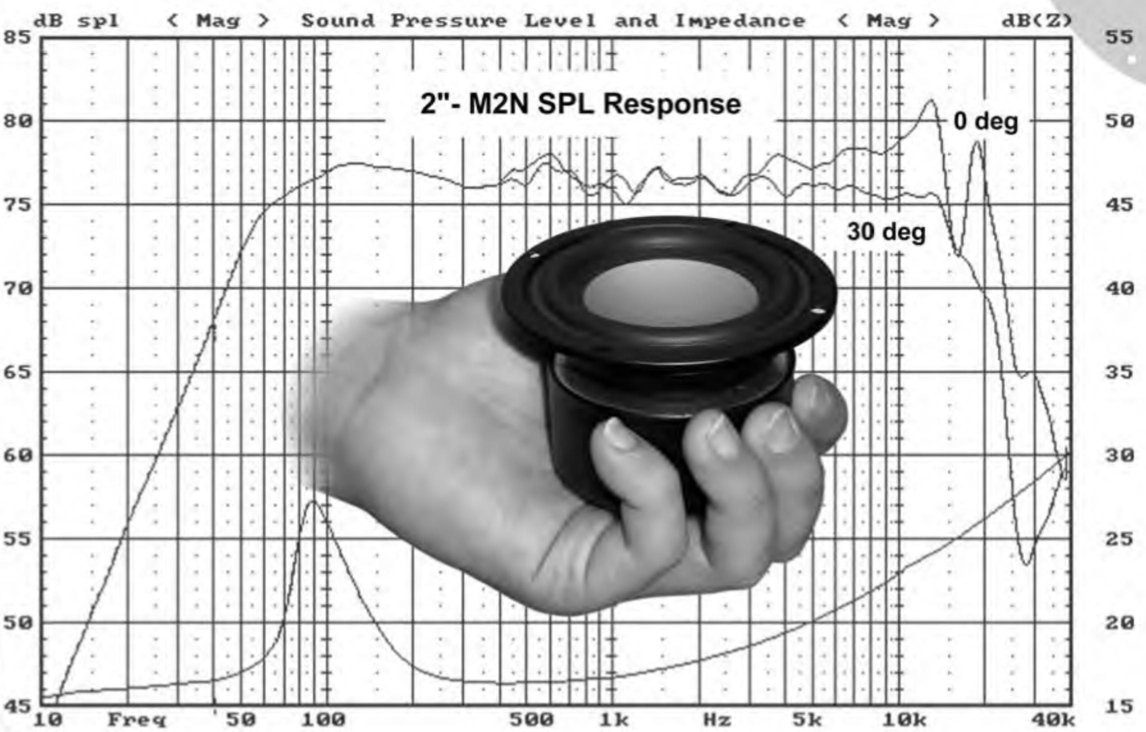


PHOTO 1: Test box showing the frame and toggle clamps for holding a test panel.

DIFFERENT MATERIALS

To see whether any material was superior, I started by examining particleboard, MDF, and plywood. The sound pressure level (SPL) chart for particleboard (*Fig. 1*) shows four resonance modes at 187.50Hz, 243.16Hz, 287.11Hz, and 383.79Hz. (Since the output is from an accelerometer it isn't sound pressure but it is useful to think of the vibrations as such, and thus I will refer to the converted impulse information as SPL output.) There are lesser modes at between 750Hz and 931Hz, which are about 10–20dB lower than the main four resonances. There seem to be some minor modes below 100Hz, but they don't appear to be a problem as indicated by the cumulative spectral decay (CSD) or waterfall chart (*Fig. 2*). The CSD shows that the resonance at 243.16Hz, though, takes well over 40ms to decay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Hz
5k
10k
40k

or decline in level by more than 20dB, which suggests the second mode is the primary resonance mode of the panel.

MDF appears to behave similarly to particleboard (*Fig. 3*). The four resonance modes are at the same frequency, but peaks on the first three are 1-2dB lower. The waterfall chart (*Fig. 4*) shows no significant differences, either.

Birch-faced seven-ply plywood shows more differences compared to MDF and particleboard. This is expected because it is stiffer than the other two. While the first three resonances are at the same frequency, the fourth is at 351.56Hz but is about 3–4dB lower. However, it now has two secondary modes at around 650Hz and 850Hz instead of the single broad mode between 750Hz and 931Hz. The CSD chart suggests a slightly longer decay time for the primary resonance mode at 287.11Hz (*Figs. 5* and *6*).

These results suggest none of the three materials tested to be significantly better than the other. However, since ¾″ MDF is the material of choice for the loudspeaker industry, I've chosen to concentrate the rest of this study on this particular material.

TRANSMISSION OF VIBRATIONS

I assumed that most of the vibration from the woofer is transmitted to the cabinet by contact or mechanical conduction. However, it occurred to me that some of the vibration is conducted through the air. To see how much of the vibration is transmitted by air, I stuffed the enclosure with 2 lb of Acousta-stuf. *Figure 7* shows

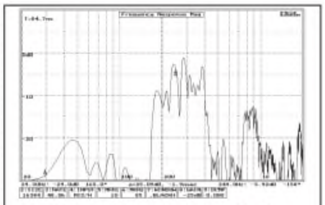
FIGURE 1: SPL of ¾″ particleboard without damping treatment.

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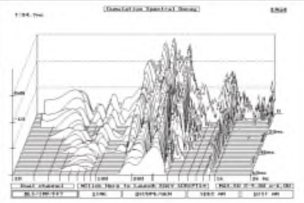
FIGURE 2: CSD of ¾″ particleboard without damping treatment.

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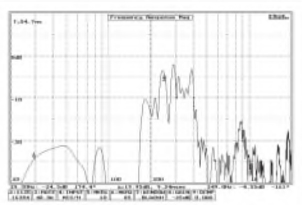
FIGURE 3: SPL of ¾″ MDF without damping treatment.

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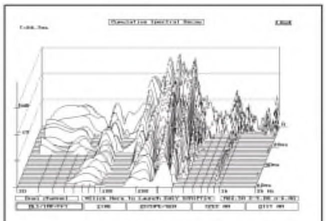
FIGURE 4: CSD of ¾″ MDF without damping treatment.

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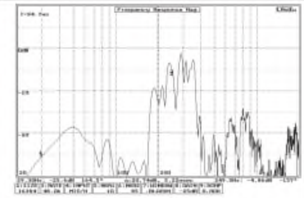
FIGURE 5: SPL of ¾″ birch-faced seven-ply plywood without damping treatment.

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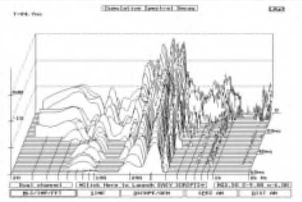
FIGURE 6: CSD of ¾″ birch-faced seven-ply plywood without damping treatment.

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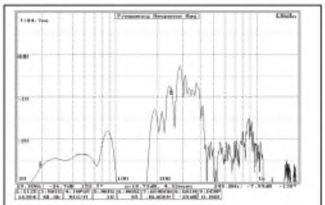
FIGURE 7: SPL of ¾″ MDF with enclosure stuffed with 2 lbs Acousta-Stuf.

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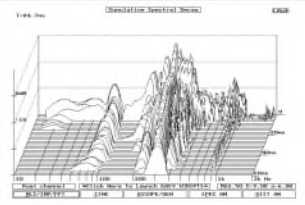
FIGURE 8: CSD of ¾″ MDF with enclosure stuffed with 2 lbs Acousta-Stuf.

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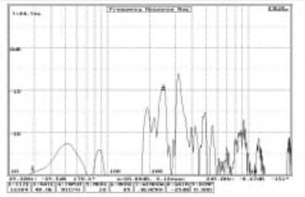
FIGURE 9: SPL of two layers of ¾″ MDF without damping treatment.

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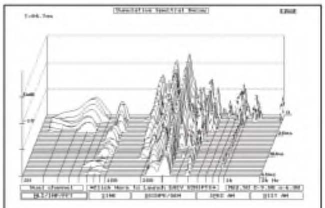
FIGURE 10: CSD of two layers of ¾″ MDF without damping treatment.

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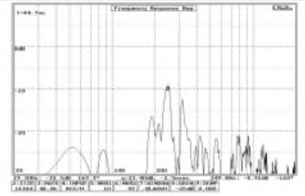
FIGURE 11: SPL of three layers of ¾″ MDF without damping treatment.

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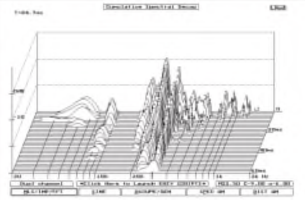
FIGURE 12: CSD of three layers of ¾″ MDF without damping treatment.

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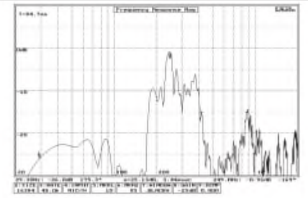
FIGURE 13: SPL of ¾″ MDF with 1/16″ BVD.

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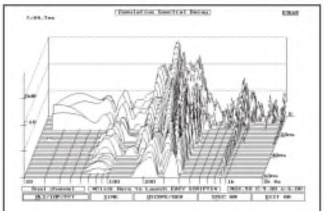
FIGURE 14: CSD of ¾″ MDF with 1/16″ BVD.

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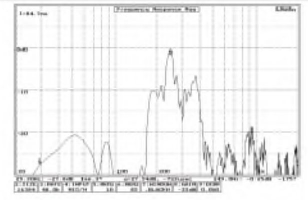
FIGURE 15: SPL of ¾″ MDF with Black Hole 5.

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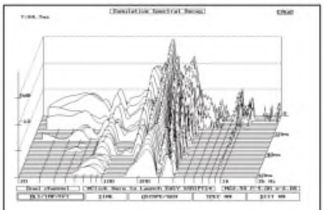
FIGURE 16: CSD of ¾″ MDF with Black Hole 5.

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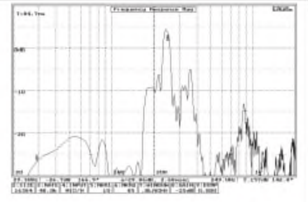
FIGURE 17: SPL of ¾″ MDF with Deflex subwoofer panel.

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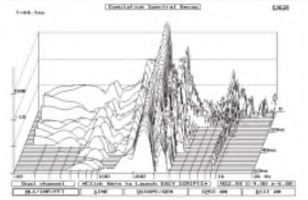
FIGURE 18: CSD of ¾″ MDF with Deflex subwoofer panel.

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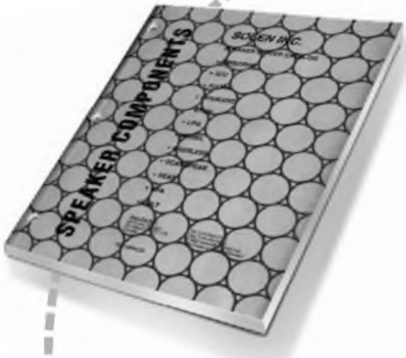
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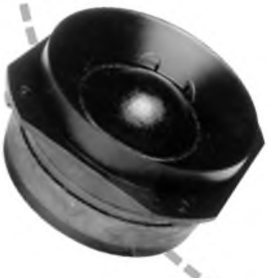


CROSSOVER COMPONENTS

a reduction in three of the four resonance peaks by 2-3dB. The waterfall chart (*Fig. 8*) confirms the modest improvement in decay.

This suggests that some of the panel vibration is transmitted by air with most of it caused by mechanical conduction. This would suggest that sealed box designs, which tend to have most of their enclosure volume filled with damping material, should be a little less affected by these resonances.

MULTIPLE LAYERS

One obvious approach to dealing with the panel resonance problem is to increase the panel thickness, making it stiffer but heavier. For example, kits from Zalytron, according to reviews in *Speaker Builder*, have cabinets built with a double layer of ¾″ MDF. The SPL chart in *Fig. 9* shows why this technique is popular; the first two modes are at the same frequency but are attenuated by 3–5dB compared to a single layer of MDF. The third mode is pushed up to 322.27Hz and is reduced by 4–5dB. The fourth mode is pushed out to 483.40Hz and is down more than 10dB. The CSD chart shows that decay time is approximately the same, however, as seen in *Fig. 10*.

Figures 11 and *12* show further gains from tripling the thickness: the first two resonance modes remain at the same frequency but are reduced by 5-6dB. The third mode is somewhere between 300-322Hz and is down more than 12dB. It isn't clear where the fourth mode is, but since the resonances above 400Hz are down by more than 20dB, they probably aren't much of a consequence. The CSD, though, still shows decay times re-

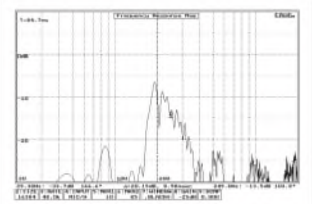

FIGURE 19: SPL of ¾″ MDF with 1″ sand-filled panel.

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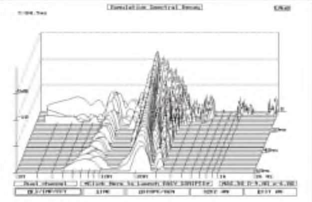

FIGURE 20: CSD of ¾″ MDF with 1″ sand-filled panel.

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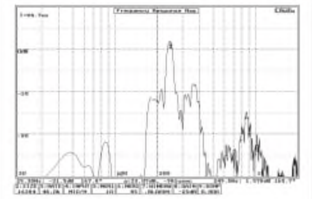

FIGURE 21: SPL of ¾″ MDF with 1⁄16″ lead sheet glued with contact cement.

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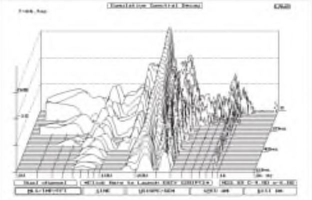

FIGURE 22: CSD of ¾″ MDF with 1⁄16″ lead sheet glued with contact cement.

B-2043-22

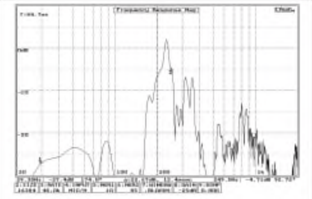

FIGURE 23: SPL of ¾″ MDF with 1⁄16″ lead sheet with ⅛″ Sorbothane.

B-2043-23

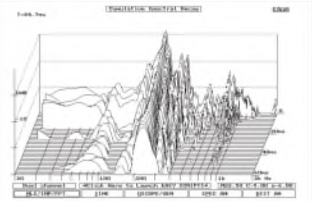

FIGURE 24: CSD of ¾″ MDF with 1⁄16″ lead sheet with ⅛″ Sorbothane.

B-2043-24

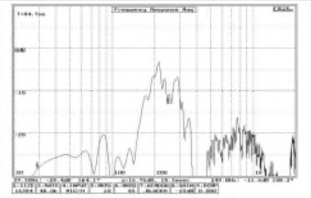

FIGURE 25: SPL of ¾″ MDF with 1⁄16″ lead sheet with ½″ Sorbothane.

B-2043-25

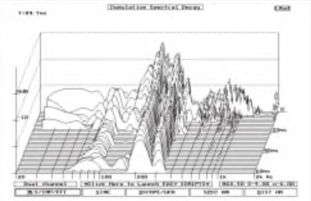

FIGURE 26: CSD of ¾″ MDF with 1⁄16″ lead sheet with ½″ Sorbothane.

B-2043-26

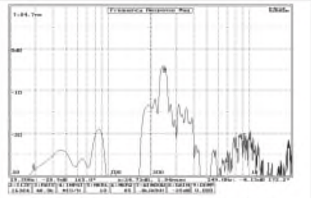

FIGURE 27: SPL of ¾″ MDF and ⅛″ Isodamp C-1002 and ¾″ MDF CLD panel.

B-2043-27

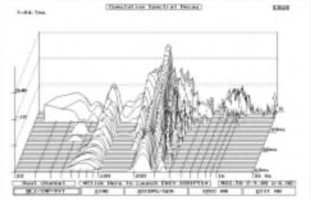

FIGURE 28: CSD of ¾″ MDF and ⅛″ Isodamp C-1002 and ¾″ MDF CLD panel.

B-2043-28

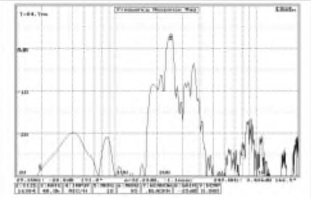

FIGURE 29: SPL of ¾″ MDF and ¼″ Isodamp C-1002 and ¼″ ACX plywood CLD panel.

B-2043-29

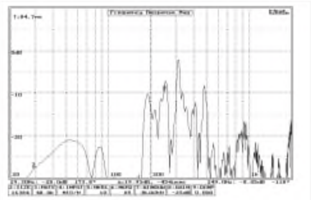

FIGURE 30: SPL of ¾″ MDF and ¼″ ACX plywood CLD panel.

B-2043-30

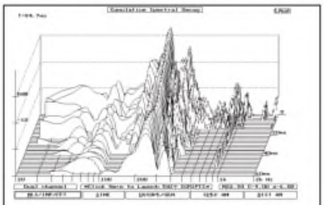

FIGURE 31: CSD of ¾″ MDF and ¼″ Isodamp C-1002 and ¼″ ACX plywood CLD panel.

B-2043-31

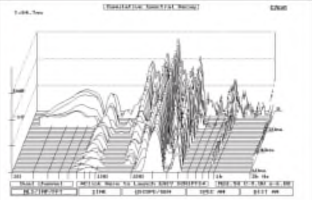

FIGURE 32: CSD of ¾″ MDF and ¼″ ACX plywood CLD panel.

B-2043-32

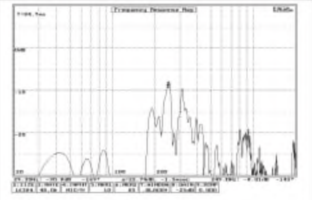

FIGURE 33: SPL of ¾″ MDF and North Creek soft glue and ¾″ MDF CLD panel.

B-2043-33

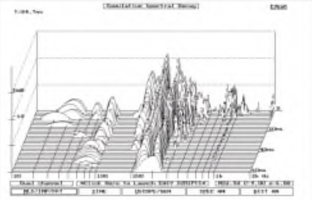

FIGURE 34: CSD of ¾″ MDF and North Creek soft glue and ¾″ MDF CLD panel.

B-2043-34

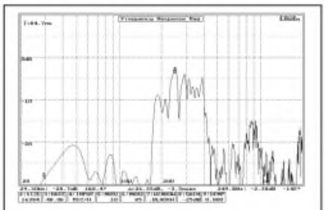

FIGURE 35: SPL of ½″ MDF and North Creek soft glue and ¼″ MDF CLD panel.

B-2043-35

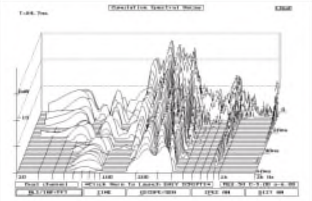

FIGURE 36: CSD of ½″ MDF and North Creek soft glue and ¼″ MDF CLD panel.

B-2043-36

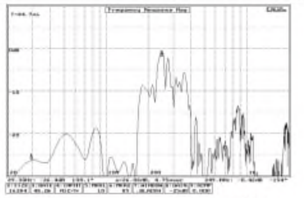
FIGURE 37: SPL of ½″ MDF and 1⁄16″ SBR rubber and ¼″ MDF CLD panel.

B-2043-37

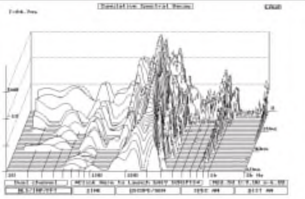
FIGURE 38: CSD of ½″ MDF and 1⁄16″ SBR rubber and ¼″ MDF CLD panel.

B-2043-38

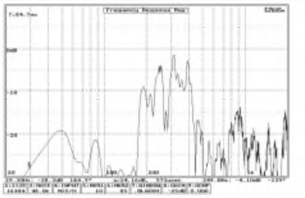
FIGURE 39: SPL of ½″ Apple-Ply and North Creek soft glue and ¼″ MDF CLD panel.

B-2043-39

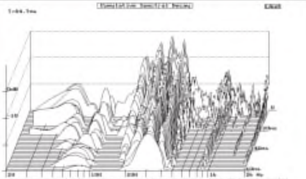
FIGURE 40: CSD of ½″ Apple-Ply and North Creek soft glue and ¼″ MDF CLD panel.

B-2043-40

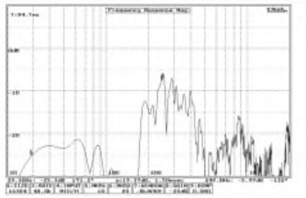
FIGURE 41: SPL of ½″ Baltic birch and North Creek soft glue and ½″ MDF CLD panel.

B-2043-41

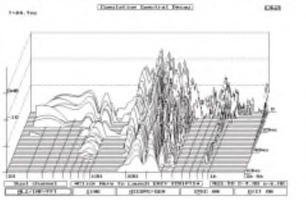
FIGURE 42: CSD of ½″ Baltic birch and North Creek soft glue and ½″ MDF CLD panel.

B-2043-42

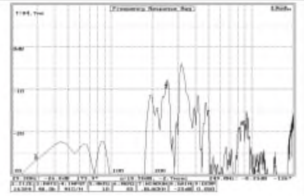
FIGURE 43: SPL of ½″ birch ply and North Creek soft glue and ½″ Baltic Birch CLD panel.

B-2043-43

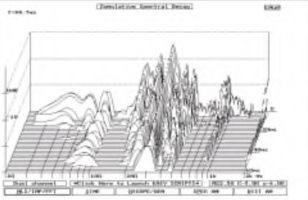
FIGURE 44: CSD of ½″ birch ply and North Creek soft glue and ½″ Baltic Birch CLD panel.

B-2043-44

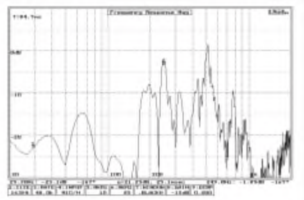
FIGURE 45: SPL of external cabinet with no brace.

B-2043-45

main quite long, at more than 40ms.

EXTENSIONAL DAMPING

This approach to panel damping applies damping material to one surface of the panel. It is also known as free-layer damping. I did an informal survey of past projects featured in *Speaker Builder*, and it seems that every builder has his/her favorite method or "recipe" for extensional damping.

For example, one project used a roofing compound, while another used a mixture of sand with yellow glue. British designers popularized the use of bituminous or tar-impregnated felt panels. In my college days in the '70s I ordered a kit from Falcon Acoustics that used such pads. Loudspeaker parts suppliers and auto sound dealers often sell an asphalt-based pad that has a self-adhesive layer. A good example of this type of material is BVD from Meniscus, which is sold in 1⁄16″ thickness.

Figure 13 shows that the first two modes remain at the same frequency but are actually 2–3dB higher! The third mode is reduced by about 7dB and the fourth mode by 3–4dB. The waterfall chart (*Fig. 14*) shows decay time for the primary mode to be extended compared to untreated ¾″ MDF.

Another material offered by suppliers is Black Hole 5, which is made up of five layers. The first is a high loss vibration damping material; the second is made of a ¼″ polyester urethane flexible open cell foam. The third layer is an ⅛″ barrier septum made of limp vinyl copolymer loaded with non-lead inorganic fillers; the fourth is 1″ polyester urethane foam; and the fifth is a thin diamond pattern embossing with polyurethane surface.

The SPL chart (*Fig. 15*) shows the first two resonances are about the same, while the third mode is reduced by 10dB, and the fourth mode by 3–4dB. The CSD chart in *Fig. 16* shows the long decay of the primary mode remains intact.

Another advertised damping material is Deflex from Spectra Dynamics. Their website states: “Made from an advanced polymer to reduce unwanted cabinet distortions to an absolute minimum. Deflex Panels control the energy, not absorb it, thus enhancing the performance of the system.”

I tested their subwoofer panel, which is larger than their other panels. This material is heavier than the previous two, and may be the cause of the increase in the

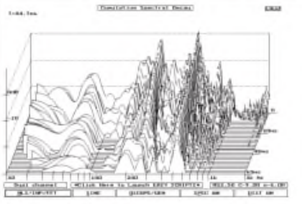

FIGURE 46: CSD of external cabinet with no brace.

B-2043-46

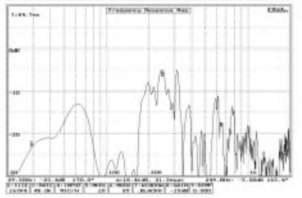

FIGURE 47: SPL of external cabinet with 1⅛″ hardwood dowel.

B-2043-47

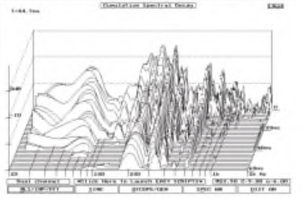

FIGURE 48: CSD of external cabinet with 1⅛″ hardwood dowel.

B-2043-48

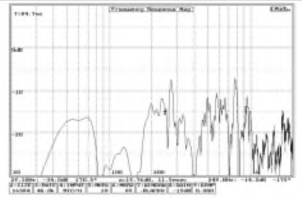

FIGURE 49: SPL of external cabinet with ¾″ × 2½ × 10½″ MDF brace.

B-2043-49

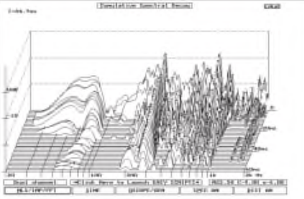

FIGURE 50: CSD of external cabinet with ¾″ × 2½ × 10½″ MDF brace.

B-2043-50

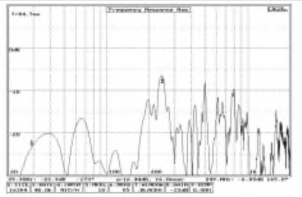

FIGURE 51: SPL of external cabinet with shelf brace with four windows and 1″ wide frames.

B-2043-51

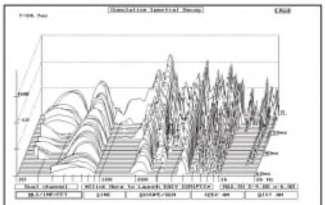

FIGURE 52: CSD of external cabinet with shelf brace with four windows and 1″ wide frames.

B-2043-52

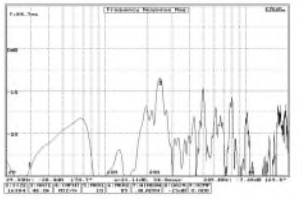

FIGURE 53: SPL of external cabinet with shelf brace with four ovals and 1″ wide frames.

B-2043-53

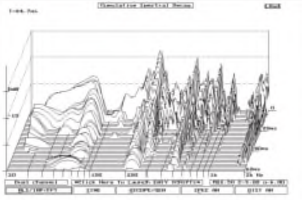

FIGURE 54: CSD of external cabinet with shelf brace with four ovals and 1″ wide frames.

B-2043-54

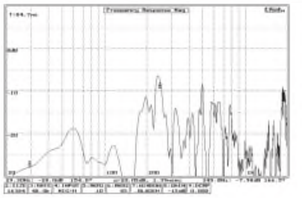

FIGURE 55: SPL of external cabinet with shelf brace with one big oval and 1″ wide sides.

B-2043-55

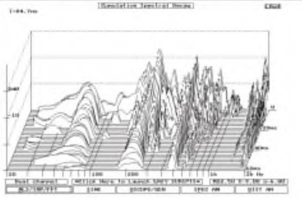

FIGURE 56: CSD of external cabinet with shelf brace with one big oval and 1″ wide sides.

B-2043-56

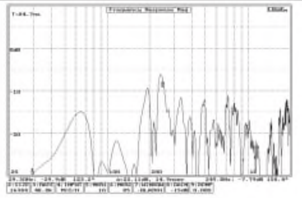

FIGURE 57: SPL of external cabinet with shelf brace with one big mahogany oval and 1″ wide sides.

B-2043-57

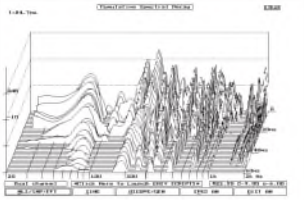

FIGURE 58: CSD of external cabinet with shelf brace with one big mahogany oval and 1″ wide sides.

B-2043-58

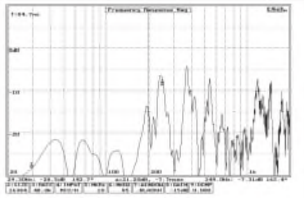

FIGURE 59: SPL of external cabinet with three shelf braces with one oval and 1″ wide sides.

B-2043-59

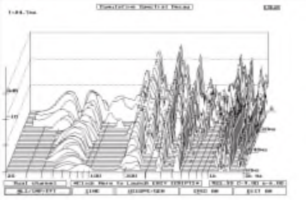

FIGURE 60: CSD of external cabinet with three shelf braces with one oval and 1″ wide sides.

B-2043-60

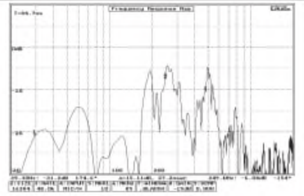

FIGURE 61: SPL of external cabinet with ¾″ SBR shelf brace with four circles and 1″ wide sides.

B-2043-61

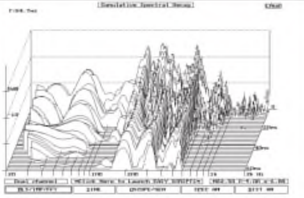

FIGURE 62: CSD of external cabinet with ¾″ SBR shelf brace with four circles and 1″ wide sides.

B-2043-62

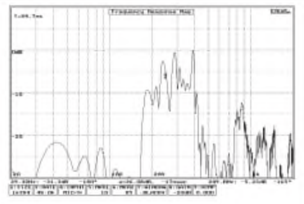

FIGURE 63: SPL of woofer with 1⁄32″ cork/neoprene gasket.

B-2043-63

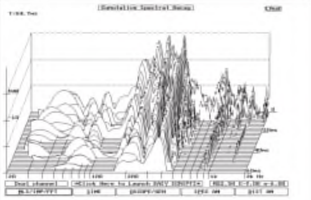

FIGURE 64: CSD of woofer with 1/32″ cork/neoprene gasket.

B-2043-64

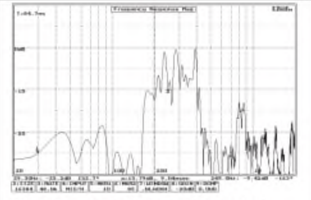

FIGURE 67: SPL of woofer with 1/16″ 30 durometer neoprene gasket.

B-2043-67

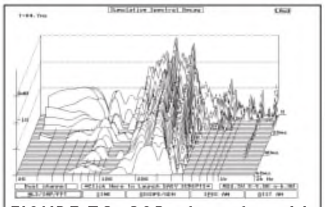

FIGURE 70: CSD of woofer with 1/4″ 30 durometer neoprene gasket.

B-2043-70

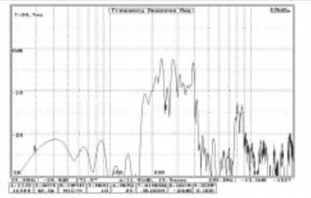

FIGURE 65: SPL of woofer with 1/8″ neoprene/EPDM/SBR foam gasket.

B-2043-65

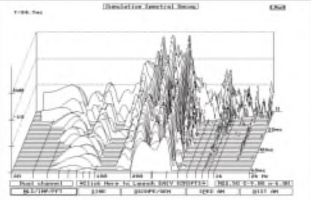

FIGURE 68: CSD of woofer with 1/16″ 30 durometer neoprene gasket.

B-2043-68

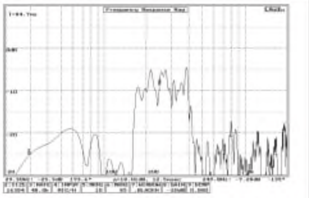

FIGURE 71: SPL of woofer with 1/16″ 30 durometer neoprene gasket and Wellnut.

B-2043-71

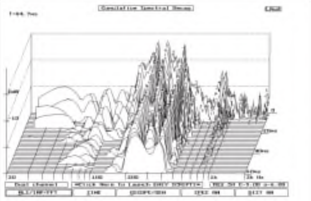

FIGURE 66: CSD of woofer with 1/8″ neoprene/EPDM/SBR foam gasket.

B-2043-66

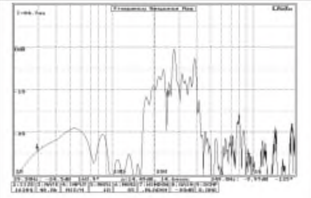

FIGURE 69: SPL of woofer with 1/4″ 30 durometer neoprene gasket.

B-2043-69

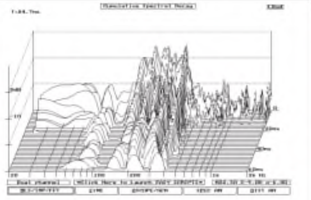

FIGURE 72: CSD of woofer with 1/16″ 30 durometer neoprene gasket and Wellnut.

B-2043-72

primary resonance mode by about 6-7dB (*Fig. 17*). The third mode is reduced more than 10dB, and the fourth mode by 3-4dB. The CSD chart (*Fig. 18*) shows the increased decay time of the primary mode with reduction of the third and fourth modes.

Next, I tested a sand-filled panel since it is considered to be a "classic" because of its apparent effectiveness. In this instance, I constructed this panel by applying 1″ of sand held in place with 1/4″ MDF panels. You can see in *Fig. 19* that the first mode is actually about 4dB higher than untreated MDF. The primary mode is shifted to 222.66Hz and is down by about 2dB. The two other modes are significantly reduced by more than 10dB. A look at the CSD chart in *Fig. 20* shows the stubborn persistence of the primary resonance mode, however.

For the past several years, I considered lead sheeting to be the ultimate damping material because of its weight and used it in several designs. It seemed ideal because it was relatively cheap and available at the local roofing supply store and did not take up much space. And it seemed to do well on the knuckle rap test.

However, it now appears that the weight of 1/16″ lead sheeting can be a help and a hindrance. A look at the SPL chart (*Fig. 21*) shows the first mode is reduced by 1dB, but the second mode is raised by about 6dB. The third mode is down by 15dB and the fourth reduced by about 4dB. And, as you can see in *Fig. 22*, the CSD chart shows the persistence of the primary resonance mode.

Figures 23–26 show the results of bonding a layer of Sorbothane, a polyether-base polyurethane viscoelastic material, between the lead and the MDF. I had hoped the Sorbothane would help damp the vibrations. While the effect of the ½″ Sorbothane is better than the 1/8″ version, the CSD chart (*Fig. 26*) again illuminates the persistence of the primary resonance at 222.66Hz. I substituted solid neoprene rubber, foam neoprene rubber, and styrene-butadiene rubber sheets of varying thickness and hardness for Sorbothane; they produced similar but less satisfactory results.

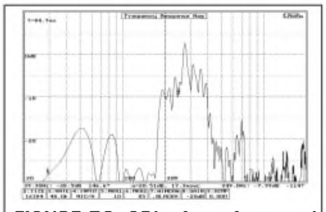

FIGURE 73: SPL of woofer panel on 1/16″ 30 durometer neoprene, 1/32″ cork/neoprene gasket.

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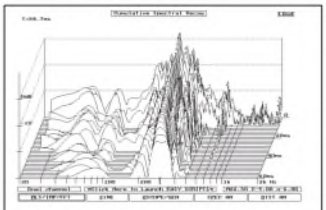

FIGURE 74: CSD of woofer panel on 1/16″ 30 durometer neoprene, 1/32″ cork/neoprene gasket.

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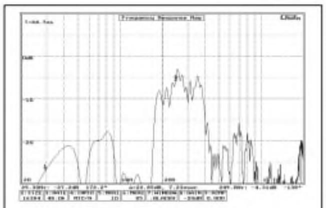

FIGURE 75: SPL of woofer panel on ¼″ 30 durometer neoprene, 1/32″ cork/neoprene gasket.

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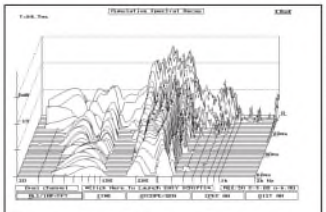

FIGURE 76: CSD of woofer panel on ¼″ 30 durometer neoprene, 1/32″ cork/neoprene gasket.

B-2043-76

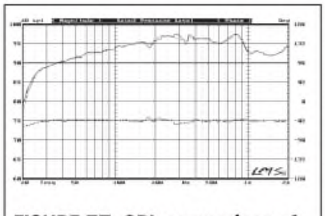

FIGURE 77: SPL comparison of ¾″ MDF and ¾″ MDF/sand-filled panel; difference curve raised by 75dB.

B-2043-77

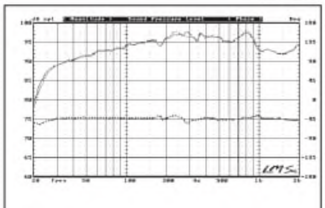

FIGURE 78: SPL comparison of ¾″ MDF and triple layer of ¾″ MDF; difference curve raised by 75dB.

B-2043-78

The difficulty of reducing this resonance suggests that most methods of extensional damping are capable of reducing secondary resonances but are ineffective when dealing with the primary resonance mode. In fact, most materials, because of their weight, appear to magnify the primary resonance. Apparently, this resonance behaves more like a weight suspended from a spring; increasing the weight increases the amplitude of the oscillation. The secondary modes may be easier to damp because their energy is distributed across a greater area.

CONSTRAINED LAYER DAMPING

Constrained layer damping (CLD) starts with extensional damping and improves it by bonding another panel to the damping material. The additional panel is called the constraining layer because it constrains the damping material. It is usually thinner than the panel being damped. Under excitation the panels move and slip thus causing a shearing force in the damping material. That is why this method is supposed to be more effective than extensional damping. You can find a more in-depth discussion on CLD at the EAR Specialty Composites website: www.earsc.com. Look under the engineering section for technical white papers. Also, a rather enlightening study on panel damping that was done by Nokia engineer Juha Backman[2] has an informative description of CLD.

I ordered samples of Isodamp C-1002, a vinyl thermoplastic produced by EAR Specialty Composites. It is used in the Sony SS-M9 loudspeaker, which was designed by Dan Anagnos and is covered by U.S. patent #5,949,033. Check it out at the United States Patent and Trademark Office: www.uspto.gov/.

The Sony speaker uses CLD for all of its exterior panels. The CLD panel comprises two 25mm panels with a 6.4mm constrained layer of Isodamp C-1002. Since the Sony speaker used two panels of similar thickness, I bonded a 1/8″ sheet of Isodamp C-1002 between two ¾″ MDF panels and tested it for resonances.

As you can see in *Fig. 27*, the primary resonance is increased by 4–5dB when compared to a double-thick layer of MDF (*Fig. 9*). The third mode is reduced by 7–8dB, but the fourth mode is higher by 3–5dB. The CSD chart (*Fig. 28*) shows the long decay of the primary mode as usual. Since this is similar to what results when lead sheeting or heavy extensional damping materials are

ABOUT DUROMETER SCALES

Durometer is the international standard for measuring the hardness of rubber, sponge rubber, plastic, and other nonmetallic materials. Very soft materials, such as gels and microcellular foam and sponge, are rated on the "Shore 00" scale. For example, chewing gum is about 20 durometer and a racquet ball about 35 durometer on this scale.

Rubber, soft plastic, polyurethane, leather, and felt are measured on the "Shore A" scale. For instance, a rubber band is about 40 durometer, car tires are about 50 durometer, and a shoe heel about 70 durometer on this scale.

Hard materials, such as hard rubber, rigid PVC, nylon, acrylic, polyurethane, and ABS, are rated on the "Shore D" scale. A bowling ball, for example, is about 55 durometer on this scale.

used, it appears the damping material and constrained layer are behaving more like extensional damping. This may be because the constraining layer is too heavy.

I also tested double ¾″ MDF panels with solid Neoprene (30 and 60 durometer hardness), styrene-butadiene rubber (75 durometer), vinyl, filled vinyl, and BVD to see whether the damping material was the problem. Results were very similar, but not as effective, when compared to the CLD panels with Isodamp C-1002. So, I then moved to examining a CLD panel constructed of ½″ MDF, ¼″ Isodamp C-1002, and ¼″ ACX plywood.

Results are shown in *Fig. 29*. The first and fourth modes are the same, while the third resonance mode is reduced by about 12dB compared to untreated ¾″ MDF, but the primary resonance mode has increased by nearly 6dB. Compared to ¾″ MDF bonded to ¼″ ACX plywood (*Fig. 30*) the first mode is up by 2dB, the second (primary) mode has increased by 12dBm, the third mode is down by 5–10dB, but the fourth mode is shifted lower in frequency and is up by 5–6dB.

The CSD chart (*Fig. 31*) shows the prominent and long decay of the primary resonance. *Figure 32* shows the CSD chart for the MDF/ACX combination. Clearly, the weight of the damping material is a factor in these results.

In his study, Backman used a commercial plywood called Schauman Wisa-Phon S. It was described as two 9mm-plywood layers with a thin viscoelastic layer between them. I wasn't able to procure this material, so after some searching I decided to build panels with North Creek soft glue (NCSG) and Kapco bookbinding glue. Both glues are a white PVA that dries clear but remains flexible, and look like the glue that is used to keep a credit card in place when it is mailed. I also took a look at but did not test an Armstrong glue for wooden floors, which is another white PVA glue that remains flexible when dried.

As you can see in *Fig. 33*, the NCSG seems to be doing a modest job of damping two ¾″ MDF panels. The first two modes are unchanged, but the third mode is down by 3–

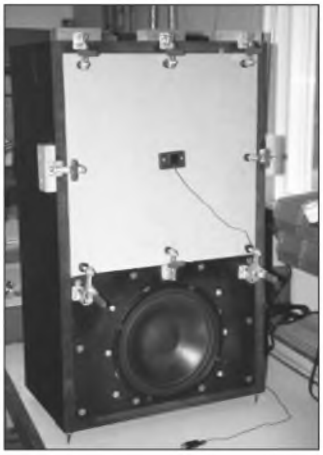

PHOTO 2: Test panel in place with ACH-01 accelerometer fastened with double-stick tape.

PHOTO 3: External cabinet with single large oval brace with 1″ frames.

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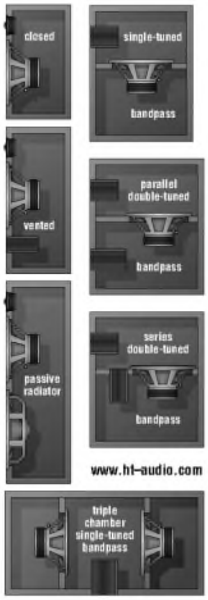

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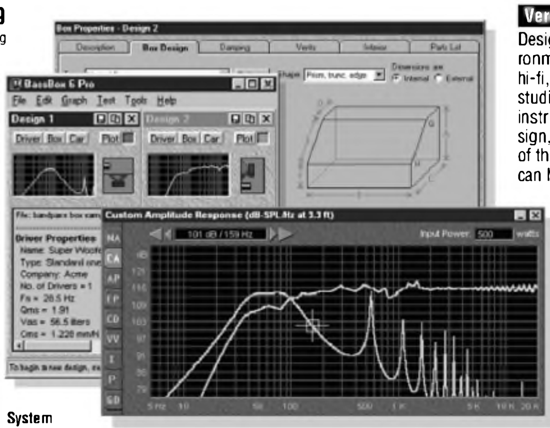

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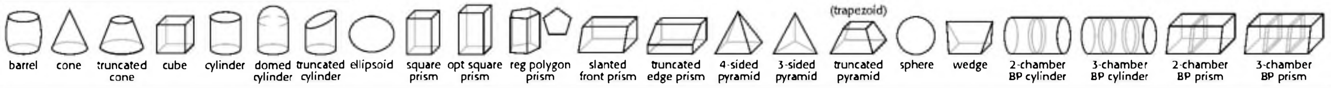

6dB, while the fourth is reduced more than 5dB. The CSD chart (*Fig. 34*) shows the decay times to be similar, however.

I then examined ways to make ¾″ CLD panels. In *Fig. 35*, ½″ MDF plus NCSG plus ¼″ MDF shows modest improvement compared to untreated ¾″ MDF. The first two modes are more or less the same, while the third and fourth modes are released by about 2dB. *Figure 36* shows a modest decrease in the decay time of the primary mode. *Figures 37* and *38* show the results of a CLD panel with a [illegible]/16″ styrene-butadiene rubber layer.

In this case the first two modes are increased by about 2dB, probably because of the weight of the damping material, while third and fourth modes are reduced by 3–5dB. The CSD chart shows a slight increase in the decay time of the primary mode.

Since increased weight seems to make the primary resonance mode worse, I then explored the use of high-grade plywood with the hope that a lighter and stiffer material would prove favorable. My favorite specialty lumberyard supplied me with ½″ nine-ply Baltic Birch and ½″ ApplePly, also a high-grade nine-ply product.

PHOTO 4: Woofer panel "floating" on ¼″ neoprene damping layer.

As you can see in *Fig. 39*, ½″ ApplePly with NCSG and ¼″ MDF produced mixed results when compared to ½″ MDF, NCSG, and ¼″ MDF. The first and fourth modes are unchanged, the second mode is down by 2dB, but the third is shifted slightly higher in frequency and increased by about 4dB. The CSD chart (*Fig. 40*) shows an increase in decay times for the major modes. A CLD panel composed of ½″ Baltic Birch, NCSG, and ½″ MDF showed good results for all four modes, however.

Compared to *Fig. 35*, all of the resonance modes in *Fig. 41* are down by 3–6dB. But, compared to double ¾″ MDF with NCSG (*Fig. 33*), modes are 2–6dB higher. The CSD chart (*Fig. 42*) shows a modest decrease in decay time of the primary mode.

Finally, a double layer of ½″ Baltic Birch with NCSG is shown in *Fig. 43*. Compared with *Fig. 35*, the ½″ MDF/NCSG/¼″ MDF sandwich, the Baltic Birch/NCSG sandwich has reduced the first two modes by 3–5dB and the fourth mode by about 10dB. The third mode, though, is pushed a little higher and is about the same level. The CSD chart (*Fig. 44*) shows a long decay for the third mode.

An examination of the various methods, multiple layers, extensional damping, and constrained layer damping suggests the following conclusions: Multiple layers can lower resonance peaks by 3–5dB and some by as much as 10dB; however, this benefit comes with the penalty of additional weight and bulk. This drawback dissuades most commercial manufacturers from employing this method. However, it is fine for do-it-yourselfers, and may be one reason why high-quality kits can produce better sound than a ready-made product. Decay times remain, however.

Most forms of extensional damping have mixed results, because they appear to only modestly reduce secondary modes but amplify the primary mode. The exception is the sand-filled panel, which is difficult to build, however. Constrained layer damping can be effective when you choose the right materials and damping materials. Double ¾″ layers of MDF and NCSG is a choice if size and weight aren't a factor.

If they are a concern, then I recommend double ½″ Baltic Birch with NCSG because it weighs nearly the same as a single layer of ¾″ MDF. In all cases, however, decay times of the resonances remain stubbornly high.

BRACING

Since it became apparent that both extensional and constrained layer damping were limited, I next turned my attention to bracing. After some thought, I devised an "external" cabinet that would allow the testing of various bracing schemes (*Photo 3*). The external cabinet is composed of two 12″ × 12″ side pieces of ¾″ MDF that I glued to the test panel and then connected with a 12″ × 10.5″ back piece. I then glued the bracing material to the test panel and the external cabinet, and installed the test panel with external cabinet for testing.

Direct comparisons with the previous tests are difficult because the external cabinet adds additional resonance modes and alters the previous modes. You can see this for a ¾″ MDF test panel with an external cabinet and without a brace in *Fig. 45*. There now appear to be two prominent modes instead of one; you can see this more clearly in the CSD chart (*Fig. 46*).

The first bracing test was with a 1[illegible]/8″ hardwood dowel, which is commonly available in three or four foot lengths from a hardware store. I glued it into place on the center of the test panel and the center of the back piece. Results with the dowel were mixed (*Fig. 47*).

Compared to the unbraced test panel the first mode is about the same, the second mode is reduced by 3dB, the third is up by 5dB. The prominent fifth mode is down by more than 10dB, however. The CSD chart in *Fig. 48* shows the impact of the increased third mode and the reduced fifth mode.

The second brace was a ¾″ × 2½″ × 10½″ piece of MDF that I glued to the center of the test panel, and glued the ends of the brace to the side pieces. As you can see in *Fig. 49*, it appears to be effective because it reduces the two major modes by 5–10dB, compared to the unbraced test panel, but is up just 2dB for one of the minor modes. The CSD chart in *Fig. 50* confirms its excellent performance.

The rest of the braces that I tested were all shelf braces; that is, they resemble shelves if you can imagine viewing the inside of the enclosure. They are not only glued to the test panel, but also to the sides and the back piece. I made most of them from ¾″ MDF, and they varied in the size, number, and shape of the holes in the shelf brace so as not to impede the flow of air within the enclosure.

The first of the shelf braces

looks like a window with panes. It includes four rectangular holes and 1″ wide frames around the "window" and the cross pieces. As you can see in *Fig. 51*, the shelf brace with four windows does a good job of lowering all of the major resonances by 5–10dB. The six minor resonances remain more or less the same. The CSD chart (*Fig. 52*) shows the reduction of the two prominent resonance modes.

The shelf brace with four oval holes also had 1″ wide frames, but the extra arch-shaped material from the oval shape added additional stiffness. *Figure 53* shows 1–3dB better reduction in peak modes compared to the shelf brace with four rectangular windows. The CSD chart (*Fig. 54*) also shows the modest improvement in the two major modes. This brace, however, is not quite as effective as the ¾″ × 2½″ × 10½″ piece of MDF. This is probably due to the frame being only 1″ wide.

While the shelf brace with four ovals or windows does very well, and could be improved by making the frame wider, there may be situations in which it would be impractical to use a brace that occupies or obstructs the central part of an enclosure. So I tested a shelf brace with a single large oval and 1″ frames. *Figure 55* shows the first mode is up by 2dB, the second by less than a dB, the third has increased by 7dB, while the others are about the same, compared to the shelf brace with four ovals. The CSD chart (*Fig. 56*) shows the modest increase in the first two modes.

The next shelf brace includes a single large oval and 1″ frame, but is made of mahogany lumber core plywood. I expected this material to be stiffer than MDF and do better; however, as *Fig. 57* shows, it produced 2–3dB increases in the first and third modes but was 4dB better on the fourth mode. The CSD chart (*Fig. 58*) shows the increase in those modes.

I then considered three shelf braces each with a single large oval but made with ¼″ MDF. I was hoping there might be some overall benefit to having more, but thinner, braces. I positioned the braces to divide the test panel into fourths. Compared to the shelf brace with four ovals, the three shelf braces with one oval made of ¼″ MDF was 1dB higher on the prominent second mode but more than 10dB up on the third mode, while the fourth was up by 3dB (*Fig. 59*). The CSD chart (*Fig. 60*) shows the increased prominence of the third and fourth modes.

Finally, I considered a resilient material, ¾″ styrene-butadiene rubber (75 durometer). This shelf brace had four circular holes, which produced 1″ frames. This material was a big disappointment (*Fig. 61*).

I thought it might be possible to dissipate some of the vibration with a non-rigid material. Compared to the shelf brace with four ovals, the first mode is up by 8dB while the others have increased by 2–4dB. The CSD chart (*Fig. 62*) shows the prominence of the first mode and the increase in the fourth mode decay time.

Apparently, bracing can reduce resonance modes by up to 5–10dB. As with multiple layers, bracing makes the panel stiffer and lowers the output of the resonance. Decay times of the resonances remain unchanged, however.

Oval or circular holes in shelf braces appear to be more effective than rectangular holes, as you would expect, because of the additional strength of the arch-shaped material. Single-hole shelf braces with side frames of 1–2½″ would be a good choice where internal space can't be compromised, such as when you must accommodate a port tube. Finally, it appears that MDF is an adequate bracing material, although you would expect stiffer materials such as plywood to do better.

Thus, a well-designed enclosure should feature extensive bracing. That is, every panel should be braced; and, if possible, double-braced by dividing the panel into fourths. Bracing is another reason why you can produce a homemade loudspeaker to outperform a store-bought one, since extensive bracing adds to the complexity, cost, and weight of an enclosure.

WOOFER ISOLATION

As demonstrated earlier, much of the vibration is caused by mechanical conduction. So finding a way to reduce the transmission of vibration from the driver to the baffle could prove effective. I started by examining the gasket that is used to ensure an airtight seal between the driver and the baffle.

One of the more common methods of sealing the woofer/baffle joint is to use foam tape, which is often available as window weather stripping at the local hardware store or is sold by various loudspeaker parts suppliers. It seems effective, but I've never believed it to be satisfactory.

So instead, I've developed the practice of cutting gaskets from 1/32″ neoprene (60 durometer) and, lately, 1/32″ cork/neoprene. These materials are available at McMaster-Carr, an industrial supply firm. Their website is: http://www.mcmaster.com/. I used to draw circles with a compass and then use scissors or a hobby knife to cut the gasket, but now I prefer to use a low-cost gasket cutter with a disposable blade. The gaskets were roughly ¾ of an inch wide.

In *Figs. 63* and *64* you see the familiar resonance peaks and long decay of the test panel with a 1/32″ cork/neoprene gasket. In *Fig. 65* the results of using a foam gasket made of ⅛″ neoprene/EPDM/SBR foam rubber are shown. Compared to the 1/32″ cork/neoprene control gasket, the first resonance mode is down by 2dB, while the second mode is moved slightly lower in frequency. The third mode is down by 3dB, and the fourth is off by about 4dB. The CSD chart (*Fig. 66*) shows an extension in decay time.

So results are mixed to modestly better for this type of material. This material does compress a fair amount; it seems to end up being about 1/32″ once the screws are hand-tightened.

The next gasket material was 1/16″ neoprene with a hardness of 30 durometer. This material appears to be similar to what is used for bicycle inner tubes. *Figure 67* shows the first mode is down by about 1dB, but the second has risen by about 2dB; the higher two modes are unchanged. Also, the mode at 80Hz is up by more than 5dB. The CSD chart (*Fig. 68*) confirms the longer decay times due to the higher levels.

A ¼″ thick 30 durometer neoprene gasket produces mixed results (*Fig. 69*). Here the first mode is lowered by 2dB, the second mode is lower in frequency and is reduced by 2dB, the third mode is unchanged, and the fourth is off by 2dB. *Figure 70* shows the still prolonged decay of the third mode.

So, it appears that a gasket made of some resilient material and of some thickness can modestly reduce these troublesome vibrations. One reason why the vibration reduction is so modest may be because the driver is transmitting vibrations through the fastening screws, which are directly connected to threaded inserts embedded in the baffle.

In his excellent *Loudspeaker Recipes* book, Vance Dickason uses Wellnuts, a brand of rubber-insulated rivet nuts, to secure the drivers and reduce the transmission of vibration. These fasteners have a rubber body with a nut at one end, which is inserted through a hole. As the nut is tightened the rubber expands to keep the fastener from being withdrawn.

In this instance, I tested a Wellnut that is 1 1/16″ long and ⅛″ in diameter and that uses a 10/32 screw. Since the Wellnut has a 1/16″ flange, I used it in conjunction with the 1/16″ 30 durometer neoprene gasket. *Figure 71* shows this combination did not lower the first mode, trimmed the second mode by 2dB, and lowered the third and fourth by 4.5–5dB. The CSD chart (*Fig. 72*) shows a corresponding reduction in decay times and a slight breakdown of the ridges in the third mode.

Because it appears that fasteners help to transmit vibrations to the panel, it occurred to me that a way to isolate the woofer baffle might prove more effective than rubber-insulated rivet nuts. So, I constructed a sandwich panel with the damping material glued between two ¾″ MDF panels (*Photo 4*).

The bottom ¾″ MDF panel is fastened to the woofer baffle with machine screws and threaded inserts. The woofer is attached to the top panel with ¾″ wood screws that do not penetrate the damping material or the bottom panel. Thus, the panel the woofer is attached to is "floating" or isolated from the rest of the enclosure by the damping material.

The first test was with 1/16″ 30 durometer neoprene. In *Fig. 73* it produced mixed results with a shift in mode frequencies, a reduction in some modes, and a significant increase in one mode (*Fig. 73*). The CSD chart (*Fig. 74*) shows the increased decay time of the prominent mode but reduced decay times for the higher modes.

With a ¼″ thick layer of 30 durometer neoprene, results were improved (*Fig. 75*). Again, the higher modes were reduced the most while the first is up a dB and the second is down a dB. The CSD chart (*Fig. 76*) shows the first-mode ridge to be intact but the others are less defined. Compared to the woofer fastened with Wellnuts and a 1/16″ 30 durometer neoprene gasket, this particular combination appears to have a slight edge in reduced decay times and slightly less prominent ridges in the CSD chart.

SOUND PRESSURE LEVEL MEASUREMENTS

All of the preceding commentary would be meaningless, of course, if these resonances had no effect on the sound we hear from a loudspeaker. Are these resonances audible? The following SPL measurements appear to definitely indicate their presence (*Fig. 77*). These are one-meter ground-plane sweeps of the test box with a ¾″ MDF panel (solid line) compared with a ¾″ MDF/sand-filled panel (dotted line).

The line at 75dB is the difference between the two curves raised by 75dB. Clearly, there are up to 1dB differences in the SPL between 150 to 400Hz and a little something between 900 and 1000Hz which coincides with the resonance modes in the ¾″ MDF panel. *Figure 78* shows the difference between ¾″ MDF and a test panel with a triple layer of MDF. Again, there are differences of up to a dB or more.

While a 1dB difference seems negligible, it is actually quite significant, indicating that the resonances are nearly as loud as the driver. A similar finding was reported by Barlow[3] who determined that the output of certain resonances approached the level achieved by the driver. This is also referenced by Colloms.[4]

These differences could easily be more than 1dB compared to a well-built enclosure, because the area of the test panel is less than a quarter of the entire enclosure. This notion is made credible by Backman's study[2], which primarily used front and rear SPL measurements of a 6½″ woofer in enclosures made from various materials. His study showed 1–2dB differences for front SPL readings between an enclosure built entirely with CLD panels compared with one made of untreated MDF. Rear SPL measurements showed the CLD enclosure to have SPL peaks that were 10–20dB lower than the MDF enclosure!

If panel-induced resonances are nearly as loud as the driver, then they must have a detrimental effect on the subjective sound of a loudspeaker. You can, without a doubt, hear these resonances in a poorly constructed enclosure.

CONCLUSION

When I started this odyssey more than a year ago, I discussed some of the initial results on extensional damping with my mentor, Vance Dickason. He politely suggested I was making a Don Quixote-like attempt to solve the loudspeaker enclosure vibration problem with that particular method. He was right, of course, because this study shows that the primary resonance seems invincible to any form of extensional damping layer. And, he was probably suggesting that multiple solutions would be required to solve the problem and that even these would only moderately attenuate resonances rather than eliminate them. Again, he was right.

This study suggests a well-built enclosure should incorporate CLD or multiple-thickness panels, extensive bracing, and some method of isolating the woofer from the enclosure. The use of rubber-insulated rivet nuts and a thick gasket might be appropriate for some designs; a "floating" woofer panel might work with a stepped front baffle approach for others.

This study, while finished for the present, resembles an unfinished book. It would have been wonderful to devise some sure-fire method, some "killer app" to kill those persistent resonance modes. Doing so might someday win someone a Nobel Prize.

For the time being, however, we are stuck with compromises. Therefore, I remain open to suggestions, criticisms, and new ideas on this subject. Please e-mail me at: jimbo@maui.net. ❖

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4. Colloms, Martin, *ibid*, pp. 280–281, Figure 7.3.

Harmony and Distortion, Part 1

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Translated from the French by Suzanne Scara. **By Jean Hiraga**

Among the different elements composing music, the harmony of sounds, of sound sequences, and of musical chords is most certainly the element of greatest importance. To human ears, the harmony of sounds is synonymous with beauty and pleasure, and its repercussions regarding your experience of psychophysical sensations are very profound.

Following the example of the harmony of shapes and colors, the harmony of sounds is a science that takes into account not only purely physical phenomena, but also man's natural reactions to a physical or acoustical phenomenon. For each of our senses—smell, touch, sight, taste, and hearing—each of these stimuli is perceived by the nervous system as a "sensation," which is not measured by one simple scale (amplitude, and so on), but instead is represented on a multitude of scales that are more or less intertwined, registering sensations such as beauty, pleasure, displeasure, cold, flatness, depth, lightness, and so forth. Certain of these sensations are also governed largely by purely physical laws. An example regarding shapes is the famous "golden number"—the relationship of proportions and dimensions that can be found in the natural realm (the shape of a leaf or of a face) as well as in the "artificial" realm (16th century paintings such as those by Luini or Leonardo da Vinci).

Without the senses of sight and hearing, light, colors, and sounds do not exist. The eyes and the ears are extremely sensitive receivers that are also affected by many nonlinear phenomena. Still, everyone can feel the mysterious harmony that exists between several frequencies whose "disparate" relationships nonetheless make up the seven basic colors of the rainbow, or even a sound octave.

Thus, when the ear hears two sounds, of different frequencies and emitted simultaneously, that "blend" together to form a "new sound" that is pleasing to the ear—or even provokes a certain emotion—these two sounds are said to be in "harmony." You could say that it is a natural phenomenon, since it could produce the same sensation in any listener.

However, you may note that to some degree the sensation of "harmony" is influenced by the factors of society, taste, culture, and historical period. As you detect numerous sound "ranges" in music down through the centuries you observe that there are numerous "chords" of sound that have thus been worked, reworked, and altered to form a "sound harmony" in conformity with the epoch in question.

For example, a chord in perfect fifth may nowadays seem less harmonious than another chord, though that other chord may be in perfect harmony in third major. So there are the questions of time period, of culture, and even of the level of progress in psychoacoustics. If, for example, you take a "harmonic" major scale or even a diatonic major scale (which progresses by natural tones), the frequency relationships between the notes are established as follows:

```
 C    D     E     F     G     A     B     C
  \  / \   / \   / \   / \   / \   / \   /
  9/8  10/9  16/15  9/8   10/9  9/8  16/15
```

You can find certain of these relationships which are said to be "in harmony"—that is, in concordance, in consonance, in affinity. The simplest relationship is ½, which corresponds to two tones separated by an octave (in which an octave corresponds to two tones, one having a frequency that is twice that of the other). This is a perfect harmony, also known as "unison" (in the same octave or in another octave). The slightly less perfect harmonies come next, such as those having a ⅔ or ¾ relationship.

Look at the example of C and G:

$$9/8 \times 10/9 \times 16/15 \times 9/8 = 3/2$$

which produces a good harmony between C and G, or else at C and F:

$$9/8 \times 10/9 \times 16/15 = 4/3$$

which produces a good harmony between C and F as well. However, if you rigorously maintain these relationships and begin the octave with the note D (in place of C), what results is a "discordant" scale. This is illustrated in *Fig. 1*, which shows these two scales. Here you see that if you take C as the note C at the beginning of a scale and compose the notes C, D, and F of this scale, you have:

$$\begin{gathered} 1 : 9/8 \\ : 9/8 \times 10/9 \times 16/15 \\ = 1 : 9/8 : 4/3 \end{gathered}$$

But if you put the D in the beginning, in place of the C which begins the scale, this time you get:

$$\begin{gathered} 9/8 : 9/8 \times 9/8 \\ : 9/8 \times 9/8 \times 10/9 \times 16/15 \\ = 9/8 : 81/64 : 3/2 \end{gathered}$$

which shows that if you have the same relationship between the C and the F, you will, on the other hand, observe a noticeable gap between the C and the

D. As this problem occurs with many notes in each octave, an infinite number of keyboards would be required in order to keep the relationships between them in harmony.

Currently, we utilize the "chromatic" scale, which consists of 12 sounds in the tempered scale. Although it is the most common scale, it still presents many difficulties that are yet to be resolved.

The so-called tempered scale is designated as follows:

C	D	E	F	G	A	B	C
1	$2^{2/12}$	$2^{4/12}$	$2^{5/12}$	$2^{7/12}$	$2^{9/12}$	$2^{11/12}$	2

(the relationship being based on the first C)

whether the $^{12}\sqrt{2}$ of a half-tone, or 1.059. Compared with the diatonic (or natural) scale—also called the "physicist's scale" or "Zarlin's scale"—you have the following:

	Diatonic Scale	Tempered Scale
C	1.000	1.000
D	1.125	1.122
E	1.250	1.260
F	1.333	1.325
G	1.500	1.498
A	1.667	1.682
B	1.875	1.888
C	2.000	2.000

You will observe, however, that the differences are relatively minor, and only an ear endowed with keen sensitivity to pitch would be able to discern any difference between these two scales. For the piano or organ, it is possible that a diatonic scale will put forth sounds that are less "dirty" than a chromatic scale. A choir may still be forced to sing on a diatonic scale in order to harmonize well with string instruments.

The orchestra conductor's "taste," or that of the first violinist, must also be taken into account, and it is well known that the famed A_3, set at 440Hz (temperature 20°C, hygrometric reading of 65%) is only rarely respected. Certain pianists as well, striving to "make an impression" on their listeners, "stretch" the octaves so much that, for the notes at the extremes, the string instruments sometimes have great difficulty tuning up.

For the tempered scale, the notes from 16.35Hz to 15,084.27Hz are indicated in the table of *Fig. 2.* At times certain pianos are tuned with a frequency-meter to nearly 1/100Hz, a practice that does not stop piano tuners or pianists from making some minor adjustments in order to make the instruments "sound better."

Speaking of this practice, it is regrettable that there is no frequency table listing the piano tuner's customary note adjustments, for such information would in turn offer a much simpler means of tuning a piano in this way. The piano is very rich in harmonics, a quality that depends on both the notes and the design of the instrument. Therefore, you would need not just one representative table, but rather a number of tables, each tabulated according to the instrument's make and model, to be at the piano tuner's disposal.

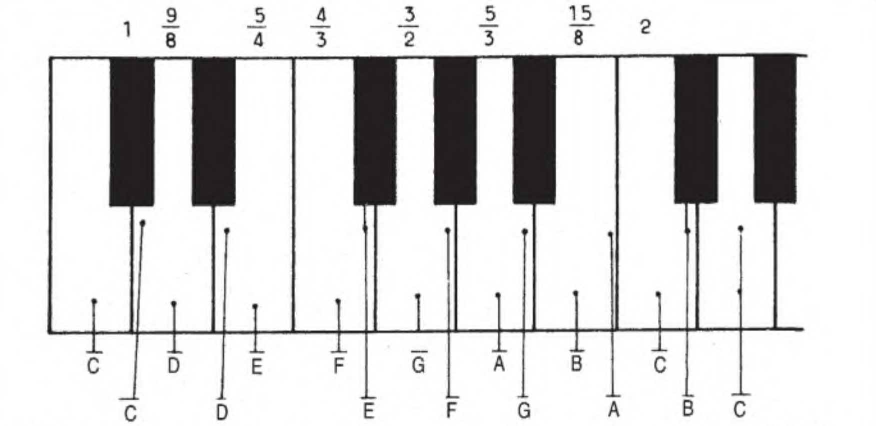
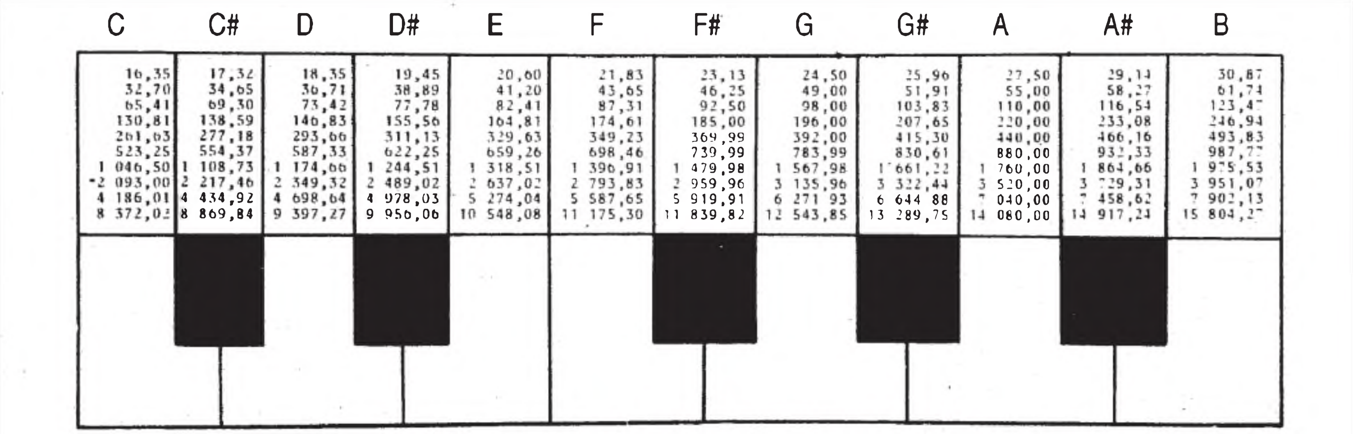


FIGURE 1: Scale in C and in D.

A-hirag-1



C	C#	D	D#	E	F	F#	G	G#	A	A#	B
16,35	17,32	18,35	19,45	20,60	21,83	23,13	24,50	25,96	27,50	29,14	30,87
32,70	34,65	36,71	38,89	41,20	43,65	46,25	49,00	51,91	55,00	58,27	61,74
65,41	69,30	73,42	77,78	82,41	87,31	92,50	98,00	103,83	110,00	116,54	123,47
130,81	138,59	146,83	155,56	164,81	174,61	185,00	196,00	207,65	220,00	233,08	246,94
261,63	277,18	293,66	311,13	329,63	349,23	369,99	392,00	415,30	440,00	466,16	493,83
523,25	554,37	587,33	622,25	659,26	698,46	739,99	783,99	830,61	880,00	932,33	987,77
1 046,50	1 108,73	1 174,66	1 244,51	1 318,51	1 396,91	1 479,98	1 567,98	1 661,22	1 760,00	1 864,66	1 975,53
2 093,00	2 217,46	2 349,32	2 489,02	2 637,02	2 793,83	2 959,96	3 135,96	3 322,44	3 520,00	3 729,31	3 951,07
4 186,01	4 434,92	4 698,64	4 978,03	5 274,04	5 587,65	5 919,91	6 271 93	6 644 88	7 040,00	7 458,62	7 902,13
8 372,02	8 869,84	9 397,27	9 956,06	10 548,08	11 175,30	11 839,82	12 543,85	13 289,75	14 080,00	14 917,24	15 804,27

FIGURE 2: Frequencies of the so-called "tempered" scale, for the notes between 16.35Hz and 15,084.27Hz. For the piano, these notes are divided up into between 27.5Hz and 4,186Hz for 88 notes.

A-hirag-2

If you simultaneously strike the notes C and G, you hear that the two sounds "blend" together well, but nonetheless they remain two coexisting notes which form a sound that "sounds" unique. Of these two notes played together, you can either hear their harmony or else "separately distinguish" the two different notes. As the opposite of "harmony" is "dissonance," you, in fact, perceive that it involves a progression from one extreme to the other, and not two separate worlds. Considering the example of light and vision, the combination of yellow and blue light rays yields the color green, a "harmonic" result in which you do not even discern the individual components (yellow and blue).

For an even more striking example, consider "white" light, such as that of the sun, the perfect harmonic resultant of the seven basic colors, so perfect that if you knew nothing but this resultant, you would be incapable of discerning its color components. Yet, for sound, it is quite another matter. With "white" noise, which you can consider "analogous" to white light, you do not perceive this noise as the perfect fusion of a myriad of pure sounds, nor as a continuous composite sound. Instead you hear a multitude of sounds of varying frequencies, which seem to change levels in a transitory fashion, and which give the impression of mixing quite badly. Thus, when you hear a composite sound, made up of many pure frequencies, you have the impression of listening not to a unique sound, but to a multitude of sounds.

The ear's ability to discern the components of this complex sound is not due to the fact that these frequencies are "too numerous" to be able to "perfectly" blend. This is due to the fact that the reproduced frequencies, infinite in number, are not quite close enough to each other, which forces the ear to distinguish separate sounds. This very important faculty of the ear was christened "Ohm's acoustic law" by the renowned German physicist Georg S. Ohm, celebrated for having discovered the basic laws of electrical current.

When two musical instruments play almost the same note, you can hear "beating," a sort of undulation of variable duration of the acoustical level, a beating that you can hear to nearly a fraction of a hertz. For this beating to disappear, and, for that matter in order to obtain two slightly different frequencies, you find that these frequencies should, in fact, be "infinitely" close to each other. In effect, "white sound" is, so to speak, almost impossible to generate, whereas "white noise" can be generated with a virtually perfect linearity.

Ohm's acoustic law demonstrates the ear's extraordinary degree of precision in distinguishing frequencies, despite a primary mechanical transmission (eardrum, middle ear, inner ear) whose mass in movement and inertia are far from negligible. For example, note that S. Asakura of the Japanese firm Grace had observed in phonographs that reading a signal of 1,000Hz could, all the same, produce an output signal of unstable frequency (and independent of

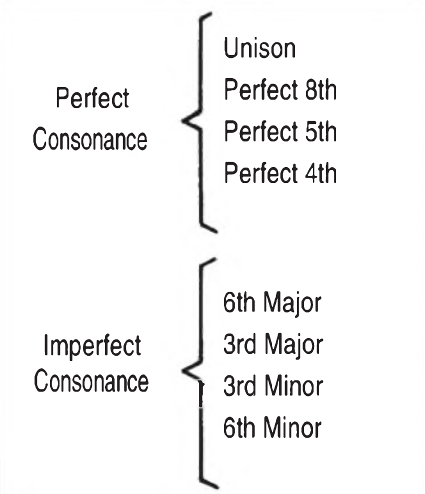


FIGURE 3: Degree of perfection of musical chords, ranked in order.

A-hirag-3

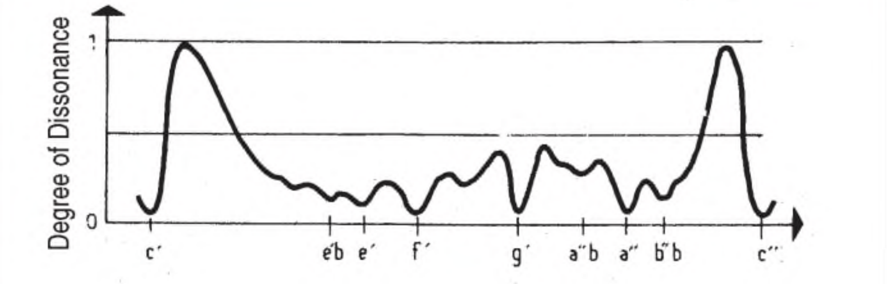


FIGURE 4: Degree of dissonance of different notes of the tempered scale played at the same time as the note C (C′, octave of the first line).

A-hirag-4

the whining or scintillating characteristics of the turntable) oscillating between 997 and 1,003Hz (effects related to dynamic mass, "trace" distortion).

As for loudspeakers, it must be even more imperfect, and all of these questions are linked to the subject of sound harmony. In a more explicit manner, Ohm's acoustic law can be verified by an experiment, which consists of analyzing the ear's possible limits concerning "critical bandwidth" (determined by Fletcher); that is, the masking effect of the white noise on some pure frequencies. Numerous physicists have studied this masking effect and, in addition to Fletcher, you could cite the works of Schafer, French, Steinberg, Zwicker, S.S. Stevens (very well-known), or J.E. Hawkins.

All this research, which may seem like abstract theoretical research in physics, can take on a much more practical appearance when applied to the reproduction of musical sounds or to the harmonic contents of an amplifier's distortion.

HARMONIC SOUNDS

When two notes an octave apart are reproduced simultaneously, they yield a new note, a new sound whose components "blend" perfectly to form a homogeneous sound. You could say that these notes "agree," that they are in "harmony." This resulting sound is a sound "harmony." By experimenting, you can establish a table of sound harmonies, ranked according to degree of perfection, and for which perfect agreement, or "unison," corresponds to a relationship of 1:1 (1.000), as shown in *Fig. 3*.

The study of harmonic sound extends back to the 15th and 16th centuries, but the most serious research is quite certainly that of the German physicist Hermann Von Helmholtz (1821–1894). He published an outstanding book on this subject (in 1862), entitled *Sensation of Tone*. From this book, you obtain a very important curve, the "harmonic (or discordant) degree" of intervals situated in an octave. *Figure 4* shows this curve, which features a scale of discordant sound measurements on one axis, and on the other, different notes played at the same time as the note C′ (C, octave of the first line). This curve shows that the "discord" between the C and the C sharp is the greatest, as with the C and the B.

You also notice that some notes are in very bad harmony with the C. The good chords are those in unison, in octave, in perfect fifth, in perfect fourth, in perfect sixth; whereas the chord in third major (relationship $2^{4/12}$ or 1.260, or 400 hundreds) yields a much less perfect harmony.

All of these questions relating to harmony, to the sound harmony's level of quality, and to chords considered "consonant" or "dissonant," are thus dependent on purely acoustical laws, which are affected by other less well-known laws, in the realm of psychoacoustics. For example, you find that, as with optical illusions, the ear itself can produce certain harmonics that are, in fact, nonexistent in the sound being heard. You can, on the other hand, produce "dissonant" chords that are particularly interesting in terms of music, or else even produce "homogeneous" and "harmonized" mixtures of dissonant chords, in which you do not sense a resulting unpleasant effect...in fact, far from it. The piece "Les dissonances" by Mozart, or even certain works by Stravinsky, prove that dissonant chord effects can yield subjective or emotional impressions that are impossible to obtain from the effect of the usual "consonant" chords.

Certain researchers (see Michel Combastet, "Physical Equivalence of Sound Timbre and Color," which appeared in *Conférences des Journées d'Etudes*, Editions Radio, 1976) have even wondered whether there was a link between harmonic sounds and "harmonic" colors; that is, a "felicitous" assortment of colors. In mathematical terms, and in terms of its progression,

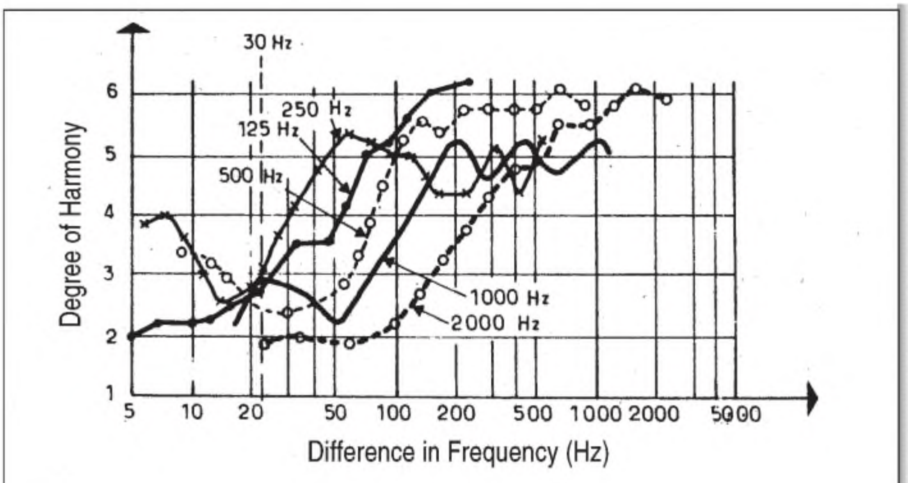


FIGURE 5: The effect of harmony or dissonance for frequencies of 125Hz, 250Hz, 500Hz, 1,000Hz, and 2,000Hz, superimposed on frequencies whose difference in frequency is between 5Hz and 2kHz. Note that an average difference of 30Hz produces the strongest dissonance, which confirms Helmholtz's theory. A-hirag-5

you perceive that the color "octave" is not at all similar to that of musical notes. Just the same, a sort of link should exist because, with sound as with colors, you are able to find a considerable variety of similar subjective impressions (for example, cool, warm, contrasting, harmonious, shrill, elegant, mellow, and so on). Certain researchers, including several of French origin, have even gone as far as producing devices capable of transcribing music into colors while attempting to respect their subjective effects.

Now, in terms of sound harmony, and provided that certain laws and standards are respected, it is possible, using arithmetic, to determine whether or not two or more sounds will agree with each other, without the help of the ear. For colors, on the other hand, it remains impossible to arithmetically determine the exact frequencies that produce a good color harmony together. At this point in time, that still seems purely subjective and, here, too, you must note an influence which is without doubt marked by the culture, the time period, and even by the prevailing fashion.

But perhaps it would one day be possible to arrive at a new science, which could be called "universal harmony," taking into account not the harmony of forms, sounds, or colors, but the "general unified harmony" shared among the five senses. Separately, these sciences are still applied only to music, architecture, sculpture, painting, photography, fragrances, cooking, or tactility (applied, for example, to the quality of fabrics). In 1981, video, a fortuitous mix of sound and color image, is therefore just a first step.

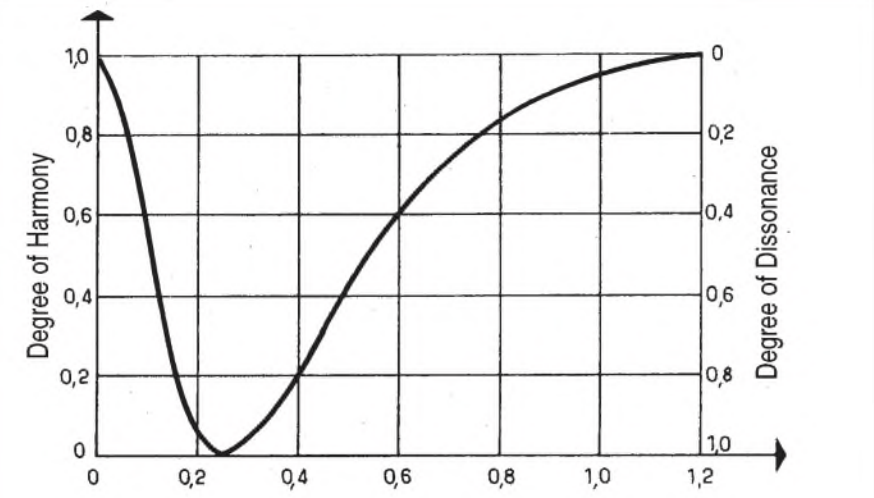


FIGURE 6: Frequency relationship between two pure notes, relating to their harmonic effect. A-hirag-6

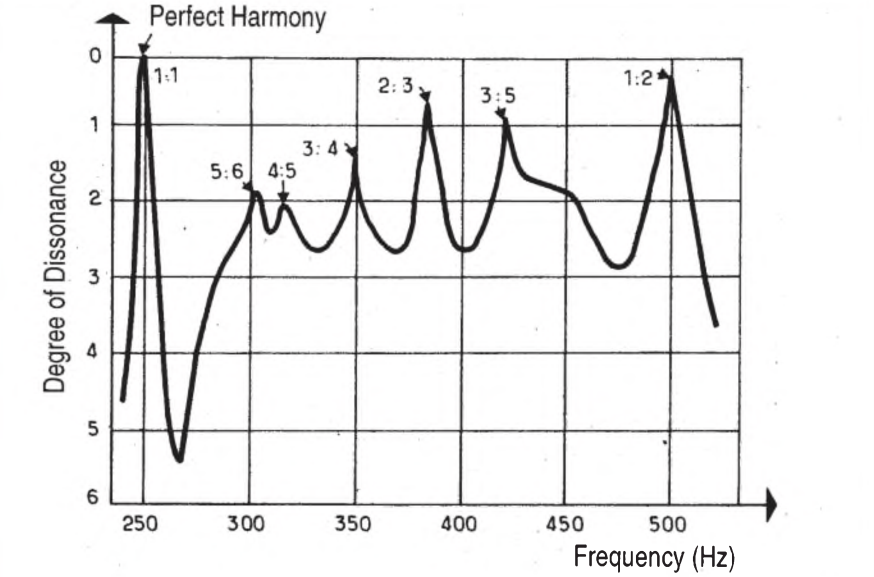


FIGURE 7: The degree of dissonance between two pure sounds, each being debased by harmonics (2 to 6). One of the complex sounds is fixed (250Hz), whereas the other slides between 250Hz and 500Hz, including its harmonics. A-hirag-7

DISSONANT SOUNDS

This subject is quite similar to that of consonant or harmonic sounds, differing essentially in that an opposite subjective effect is produced. When two pure sounds are emitted at the same time, you perceive that the degree of dissonance varies according to a particular cycle at each octave. When these sounds have exactly the same frequency (unison), you hear just one sound, the harmony being perfect.

As these frequencies begin to grow apart from each other, a "beating" of variable frequency is produced, which is to have a more or less negative effect on the quality of the composite sound. When the difference in frequency is between 20Hz and 100Hz, the dissonance is significant. If this difference in frequency grows larger, the harmony slowly returns, then disappears again as soon as the higher octave is again attained.

The works and research on dissonance which relate to psychoacoustics are relatively recent. The most interesting studies are certainly those of Kuriyakawa (Toshiba, Japan), Kameoka (Japan), and Plump (Dutch Laboratory of Psychoacoustics), all of which date from the 1960s. In a way, they complete the investigations of Von Helmholtz, according to whom a difference between two frequencies of 30Hz would produce a strong dissonance. The curves in *Fig. 5* show that it is more precisely a question of a dissonance that is also dependent on the average of these two frequencies.

Figure 6 illustrates another curve which represents the level of consonance (accord), the level of dissonance (same scale, but reversed), as well as the difference in frequency between

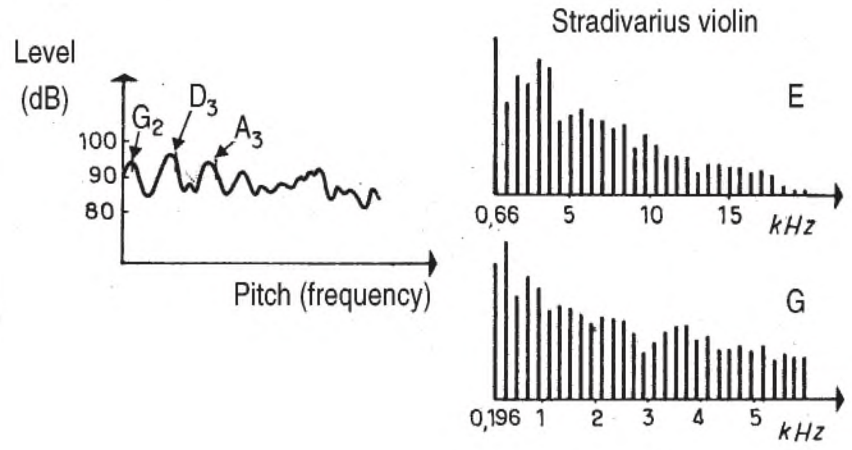


FIGURE 8: Characteristics of a good violin. Response curve, harmonic spectrum.

A-hirag-8

pure notes, called by Fletcher the "Critical difference in bandwidth." This curve, transposed on a "well-tempered" scale, illustrates that certain chords, though imperfect musically, sound "better" (i.e., have a better subjective consonance) than other more "perfect" chords. This also explains why a good composer must possess either a very acute sense of harmony or else an advanced knowledge of acoustical physics and psychoacoustics in order to make a true success of a musical work. This can even compel certain musical performers to "readjust" certain notes, even at the risk of subjecting themselves to criticism.

But you must keep in mind that the "harmonies" and "dissonances" discussed up to this point concern the combining—whether felicitous or not—of pure sounds; that is, without any harmonic distortion. This is not the case with musical instruments, and it is even less so with components designed for sound reproduction.

Suppose, for example, that an instrument simultaneously emits two frequencies—250Hz and 375Hz—chosen because they represent between them a chord in perfect fifth. Suppose that neither of these frequencies is pure, but that they also contain harmonics. The difference between these two frequencies is 125Hz, and you would thus have harmonics of frequencies equal to 500Hz, 750Hz, 1,000Hz, 1,250Hz, and 1,500Hz, respectively, on the one hand and on the other, 750Hz, 1,125Hz, 1,500Hz, and 1,825Hz.

Now suppose that one of the basic frequencies (350Hz) climbs upward and settles at 375Hz, which would yield harmonics of 770Hz, 1,155Hz, 1,540Hz, and 1,925Hz, respectively. Despite the difference in frequency increasing from 125Hz to 135Hz, you observe a uniformity of harmony if there are only basic frequencies. On the other hand, if you consider the consonance, the harmony (or dissonance) that can arise between the harmonics of each of these basic frequencies, you notice, for example, that harmonic 3 of the first frequency (750Hz) this time shows a difference of 20Hz from harmonic 2 of the second frequency (770Hz), which brings about an effect of marked dissonance. Moreover, you observe that the respective harmonics 4 and 6 (1,500Hz and 1,540Hz) present a difference of 40Hz.

These two dissonances, presented by the harmonic frequencies of these two basic sounds, show that even if the graph in *Fig. 6* indicates a good harmony, if harmonics are added the whole set of basic frequencies will yield, for even a small difference in one of the basic frequencies, a subjective effect of pronounced dissonance.

If you now add six harmonics to a pure frequency of 250Hz, you produce an acoustic mixture of this sound having the same sound, composed of the same harmonics, but capable of sliding in frequency (with a consequent slide in harmonics) between, say, 250Hz and 500Hz. Look next at the degree of dissonance between 250Hz and 500Hz (one of the composite sounds being fixed) on the frequency scale of half-tones in a tempered octave. This study is of much more practical interest because it is much closer to an actual condition.

Figure 7 shows this degree of dissonance and a strong similarity to results obtained by Von Helmholtz and other researchers in the field of psychophysics. Comparing *Figs. 6* and *7*, you can observe quite a radical change in the look of the consonance curve, which is due to the very pronounced effect of the harmonics contained in each of the basic frequencies.

Were it not for these difficulties brought about by the presence of harmonics, it is certain that this science of harmony and the design of musical instruments would be simplified considerably. *Figure 8* shows the amplitude/frequency response curve of a very good violin of Stradivarius caliber.

You notice that this curve is not linear, but presents instead for virtually all good violins three resonances (G_2, D_3, A_3), and that each of these frequencies often consists of three pronounced harmonics (dominant, mediant, tonic), thus forming for each note played an excellent "harmonic chord." But this can all become complicated if a microphone playing back these chords—good or not—is replaced by an artificial playback head or a listener.

In certain cases, a mixture of two monaural channels can bring about an effect of marked dissonance, whereas when these two channels are listened to separately via stereo headphones, there is no "dirtying" effect on the sound. In fact, there was an incident of a disturbing beating when the channels were mixed. Concerning this subject, I cite the example of a Japanese record publisher who in about 1960 had sought to correctly combine two monaural channels. In order to do so, he had to use slightly advanced tape speeds in an effort to suppress the beating effect.

We continue this study next issue, beginning with a look at harmony and "well-tempered" distortion. ❖

The AC Power Line and Audio Equipment

Part 3

We conclude our look at measures to improve and maintain AC power quality suitable for your audio equipment. **By Charles Hansen**

PHOTO 1: Power line monitor.

As I described in Parts 1 and 2 of this series (Sept. '01 and Oct. '01 *aX*), the alternating current at your AC receptacle is laden with power-line harmonics, voltage fluctuations, transients, radio interference, and other disturbances. Of all these, the surge and impulse increases in line voltage have the greatest potential (no pun intended) to damage your audio equipment.

Recall that in-phase impulses and spikes can range from 400V to 5600V, and last between 0.5 and 100µs. Lightning is one of the leading causes of these transients on utility power lines. The surge currents associated with lightning strikes interact with the distribution system impedance, producing large voltage transients that are transmitted to remote parts of the grid.

Photo 1 shows my power-line monitor[1] display after a nearby thunderstorm passed by. The surge (>127V RMS for 0.5s), spike (>200Vpk for 50µs), and outage (<100V RMS for 50ms) LEDs are all lit. While we did not experience an outage at our home, there was one noticeable brief flicker in the lights, and a touch lamp turned itself on. A week earlier, another thunderstorm took out the phone-line protection in my computer surge protector, which I had to replace. My modem and fax machine were not damaged.

SURGE PROTECTION

Figure 13 shows the schematic of my surge protector. Most consumer surge protectors, including this one, use bidirectional metal oxide varistors (MOV). It has a 15A thermal circuit breaker and an 8A non-replaceable fuse. The 100nF capacitor absorbs small spikes and RF noise. The neon light provides a "protected" indication when the unit is operating properly. Some models also provide a "grounded" light to verify that the connection to house ground is intact.

Another valuable feature of this particular surge protector is that the SPDT power switch shorts the black and white power leads of the protected outlets together when the switch is off. This significantly reduces any possibility of transient voltage across the protected equipment when this switch is off.

Another useful option (if you can find it) is an overvoltage shutoff. A sustained voltage between 121 and 184V AC, where most suppressors have little to no effect, will activate a disconnect relay until safe voltage is restored.

The circuit at the top of the schematic is the telephone-line surge protection. A sidactor is used instead of a MOV. This solid-state device is designed for telecommunication protection, and provides transient voltage clamping from the balanced tip-ring to ground. The sidactor has a much lower junction capacitance (50pF) than a MOV (1nF).

This is important because the telephone line is also a high-speed data

ABOUT THE AUTHOR

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

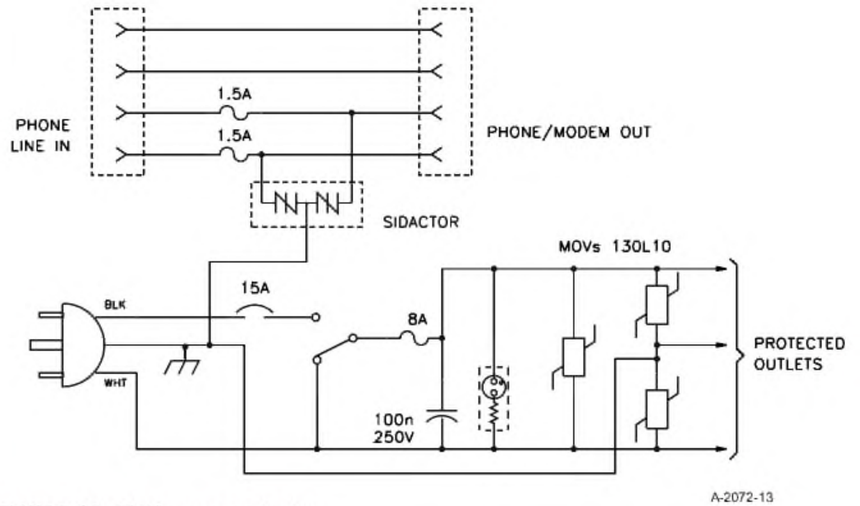


FIGURE 13: MOV surge protector.

line, and a MOV would seriously degrade the signal bandwidth. Since the sidactor has limited energy-absorption capability, 1.5A fast-blow fuses are used in series to open the tip-ring circuit to the protected telecom equipment. It was these fuses that opened in my unit.

MOVS

MOVs are composed of a thin zinc oxide disc that has a known breakdown voltage characteristic. As the line voltage approaches breakdown, the MOV begins to conduct current. At voltages slightly above the breakdown, large currents flow, effectively clamping the output voltage.

The disk diameter of a MOV determines its surge current capability. The 10mm varistor typically used in commercial surge suppressors is rated to handle a surge up to 4,500A based on the standard 8 × 20μs waveform defined by IEEE C62.41. A 20mm varistor can handle 6,500A, and varistors with surge ratings up to 40,000A are available.

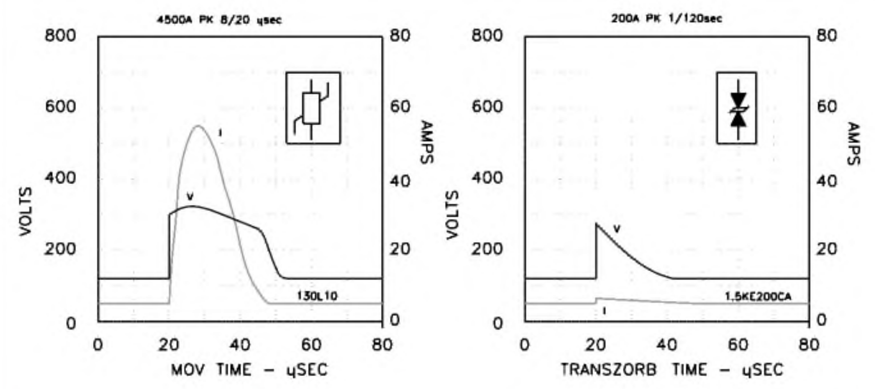


FIGURE 14: MOV versus TransZorb characteristics.

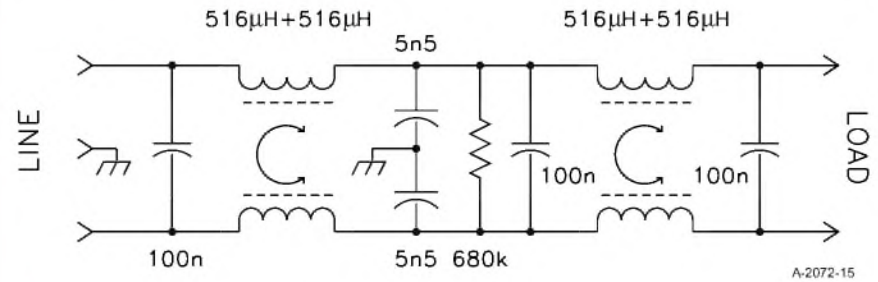


FIGURE 15: AC line EMI filter.

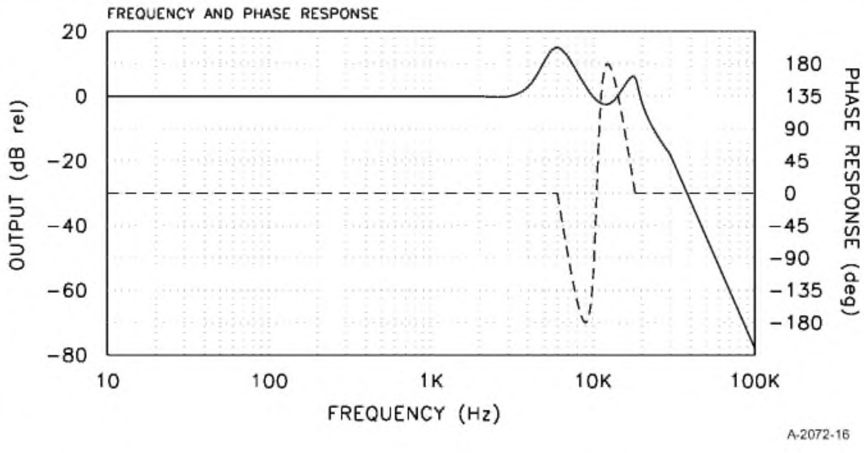


FIGURE 16: Frequency and phase response—AC line EMI filter.

Whole-house surge protectors are available for connecting directly to your service panel through one of the two-pole 220V breakers. Some manufacturers connect multiple varistors in parallel to increase the clamping level. However, due to the fact that MOVs are nonlinear, the clamping level for the varistor array will reduce the let-through voltage and compromise equal sharing of surge current among the parallel devices. The lowest impedance device will hog most of the surge current, reducing the life of the surge protector and diminishing its ability to suppress larger magnitudes of surge current.

MOVs also have an inherent wear-out mechanism. Many small impulses, or one large one, will cause degradation in the metal-oxide material. The breakdown voltage decreases and the reverse leakage current increases, which eventually leads to failure. The MOV can then overheat and even burst into flame, so a properly-sized series fuse (*Fig. 13*) should always be part of a surge protector.

For this reason, MOV surge suppressors are "one-time" devices. After a large surge, if the protected light is dark, the entire suppressor must be replaced. Some manufacturers provide insurance for damaged equipment and free replacement of the suppressor.

AVALANCHE SUPPRESSION DIODES

Another type of clamping device is the Silicon Avalanche Suppression Diode (SASD). General Semiconductor's TransZorb® is the most widely known. Instead of metal oxide, the SASD uses a silicon avalanche diode, similar to a zener diode.

The difference between a SASD and a zener is that the zener is primarily designed to handle steady-state power dissipation, while the SASD is designed to absorb high levels of energy (joules) for milliseconds. The SASD device is inherently unidirectional. Two back-to-back SASDs are required to clamp AC voltages, so bipolar SASDs are available in a single package.

SASD devices have some inherent advantages over MOVs. They have a sharper knee near the breakdown voltage. As a result, devices with ratings closer to the normal peak voltage of the AC waveform can be used. In addition, they do not degrade with repeated use, as do MOVs.

However, SASDs have relatively lower power-handling capability. The SASD has superior voltage clamping characteristics, but the MOV can handle more energy per unit volume, as you can see from the current waveforms in *Fig. 14* for two UL-approved devices.

Many SASD devices in parallel are required to provide the equivalent energy rating of one MOV. As a result, SASD-based surge suppression often costs much more than MOV-based devices. SASDs have a wider operating temperature range, and are often used in military hardware.

ELECTROMAGNETIC INTERFERENCE (EMI)

EMI is broad-spectrum interference, either conducted on the power line or radiated by wiring or circuits to susceptible equipment. Radiated EMI is composed of E-field (electric) and H-field (magnetic) components. Electric fields exist in space between two potentials, while magnetic fields exist around a current traveling through space or a conductor. Electric fields can easily be controlled with ground shielding.

Magnetic fields can penetrate ordinary conductors and require high-permeability magnetic materials to contain them. Twisted-pair wiring can help cancel the magnetic field and reduce the H-field, which is why it is used in the internal AC wiring of audio equipment.

The most common EMI radiator is the AC power cord of an electronic device. The rectifier diodes in power supplies produce spikes and ringing on the AC line as they abruptly turn off and on. Filter capacitors draw large pulses of current at the voltage peaks (see Fig. 10 in Part 2). The power cord or internal wiring may propagate any conducted interference, especially if the conductor length is close to one-quarter wavelength of the interference frequency.

POWER CONDITIONERS AND FILTERS

The purpose of power conditioners and local EMI filters is to reduce the emission of EMI (suppression) and to prevent EMI from entering the electronics (noise immunity). *Figure 15* shows the schematic for a Corcom 5VR1 EMI filter.

The 5VR1 is a six-element low-pass filter whose frequency and phase response are shown in *Fig. 16*. It has the same response in either direction. The symbol between the two halves of each inductor indicates that they are common-mode chokes. Connected to the standard 220V circuit (split into two 120V circuits), 120V equipment is unbalanced, and the neutrals and house ground are connected together. The difference (Δ) between the 4A and 8A loads in *Fig. 17* appears as a 4A ground current.

In addition, the capacitors in the EMI filter, unbalanced by the common ground and return connection, will shunt RF current into the ground. Reactive and nonlinear loads such as light dimmers also shunt harmonic currents to ground. Most ground-induced noise is directly linked to the unbalanced

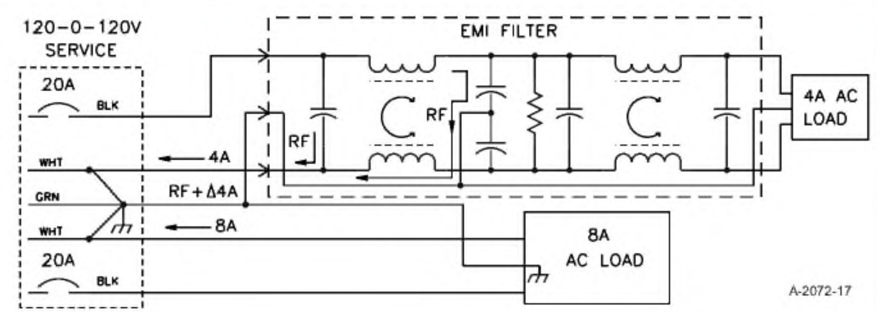
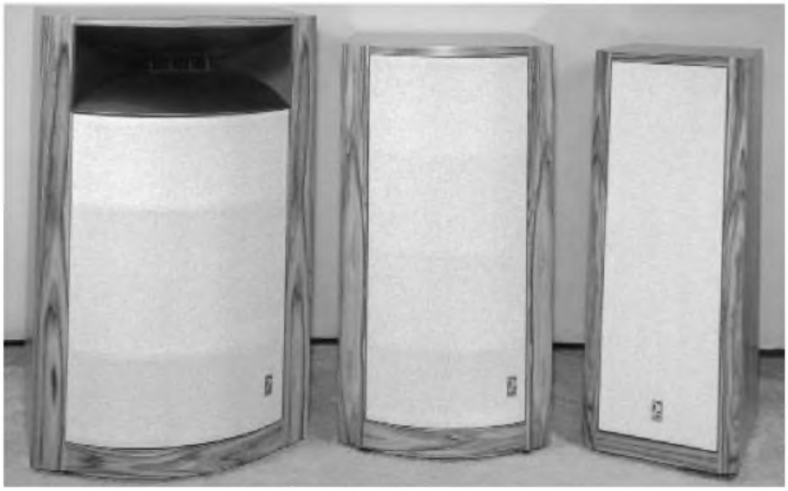


FIGURE 17: Ground current in unbalanced 120V AC circuit.



120V-0-120V split power in your home. This reduces the effectiveness of EMI filters, and may even result in more AC line noise than circuits without filters.

BALANCED POWER TRANSFORMERS

The 220V AC power line into your home is balanced for 220V loads (appliances such as electric dryers, ranges, and air conditioners), but not for the 120V AC loads. You can produce a balanced 120V AC circuit by using a balanced power-line transformer that meets Article 530-G of the National Electrical Code (NEC). This is very important to ensure safe grounding of the loads attached to the center-tapped transformer secondary (see the upper circuit in *Fig. 18*).

Figure 19 illustrates the currents in a balanced center-tapped power transformer. The currents in each 60V AC half are out-of-phase at the common ground. Both the real and reactive harmonic current components are perfectly balanced, and null to zero at ground. The load receives the full 120V AC while the transformer rejects common-mode noise, just like the balanced input transformer in a microphone preamp.

The balanced power transformer should have bifilar (two-in-hand) secondary windings to balance its two 60V AC output circuits as closely as possible, and a Faraday shield between the primary and secondary (this is the grounded dashed line in the transformer schematic). If done properly, this cancels high-frequency harmonic currents out to the low MHz. The RF current through the EMI filter caps is now balanced and nulled out.

A balanced transformer will add its own impedance in series with the equipment that it powers. This may allow some interaction in the AC power between all the components connected to the transformer, even though the common-mode noise is cancelled.

POWER REGENERATORS

While the balanced power transformer can eliminate any induced ground current, it cannot change the harmonic content of the AC line 60Hz sine wave. One method for accomplishing this is with a power regenerator (*Fig. 20*).

This is essentially a high-voltage solid-state bridged stereo amplifier, with a low-distortion sine-wave oscillator that feeds the input of the amplifier with a variable frequency (50Hz to 120Hz) reference. The amplifiers generate balanced 60-0-60V AC output power. A balanced power transformer with surge protection is used at the input to eliminate ground noise.

Commercial power regenerators are limited to 200W to 750W. This provides adequate power for lower amperage loads such as preamps, processors, CD or DVD players, and for turntables, where the variable frequency feature can also be used for speed adjustment. A higher line frequency will allow more effective operation of the audio equipment power transformers and filter capacitors, and can be used to optimize the audio noise floor.

The amplifier feedback provides voltage regulation with a claimed output impedance of around 0.01Ω, but, like all power amplifiers, the limiting factor will be the power-handling capability before clipping.

Incidentally, an uninterruptible power supply (UPS), such as the ones designed for computer backup, do not provide a clean source of power suitable for audio equipment. They are designed for maximum efficiency to extend battery operation for as long as possible. There are two types: off-line and on-line. If the AC line is the primary source and

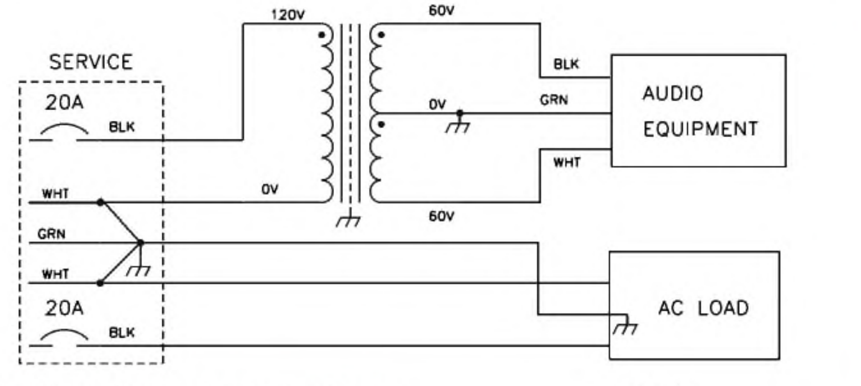


FIGURE 18: Balanced power transformer.

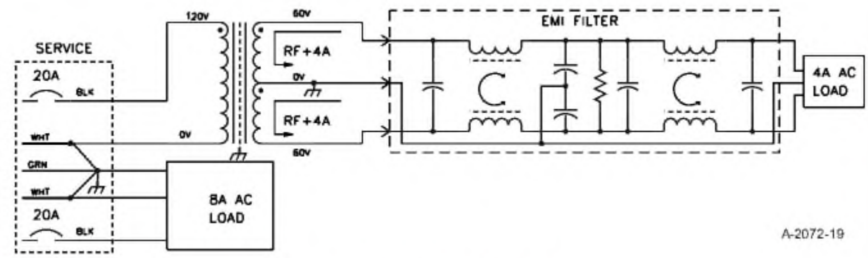


FIGURE 19: Ground current in balanced 120V AC circuit.

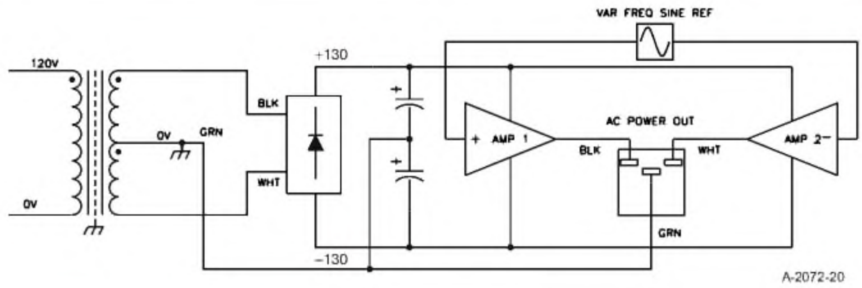


FIGURE 20: AC power regeneration.

the internal battery/inverter is the backup, the topology is known as an off-line UPS. If the UPS provides continuous inverter power from the mains, switching to battery when the mains voltage sags, it is an on-line UPS.

The impedance of the internal inverter usually results in a flat-topped output voltage waveform where the peak output current occurs. The inverter may also have power factor correction (PFC), but the power-switching circuitry can introduce high-order harmonic distortion on the output power. Since the switching power supplies in PCs are fairly tolerant of line-voltage excursions, the inverter may not switch on until the mains voltage is pretty far afield—below 90V AC or above 140V AC.

POWER CORDS

There is no question that the AC power cord of any electronic device is an EMI antenna. The power cord or internal wiring may propagate any conducted interference generated by the device, especially if the conductor length is close to one-quarter wavelength of the interference frequency. AC power cords not only act as antennae for this noise, but also conduct it back and forth between all the audio components.

However, the many feet of electrical wiring in your house is not audiophile-grade, and the usual interior wall construction materials have no shielding effect whatsoever. Many question how that last few feet of power cord can really make any difference. Therefore, audio power cords join interconnects and speaker cables as controversial high-cost items which divide audio enthusiasts into the objective/subjective camps. I don't intend to get into that here. My intent is just to describe the available technology.

Every audiophile power cord I investigated is shielded, which attenuates EMI between the wall outlet and the audio equipment. Next, the insulating materials used for each conductor have more stable dielectric properties than the rubber or PVC used in ordinary line cords. This is important in antenna design.

Teflon® is frequently mentioned (along with other undisclosed "advanced," "military-grade," "patented," "proprietary," "special," and/or "unique" materials). Much of the wire is oxygen-free copper (OFC) with silver plating, which is a standard process with Teflon insulated wire, since the high-temperature Teflon extrusion process would oxidize any exposed copper.

Some power cords use twisted-shielded construction, while others are coaxial designs. Some use one overall jacket shield; others add shields to each conductor. Some are designed to have a matched series impedance between line and return. Some are designed to have high capacitance and inductance and low resistance so the cable acts as a distributed low-pass filter. In others, either the AC plug or the IEC connector at the equipment end contains a discrete RF filter.

Some cords are designed to provide a certain ratio (5:6, or the "golden" ratio of 1.618:1) between the number of strands in the inner and outer conductors. One cord claims to have magnetic field energy storage capability to increase instantaneous current demands. Another cord is tuned to a precise two-meter length (150MHz in free space). I leave it to you to evaluate whether these properties will provide sonic improvements in your own system.

SUMMARY

Surge protection won't improve the sound of your audio system, but it may save your equipment from a large power-line transient. Power conditioners and/or power-line filters may offer sonic improvements, but they may also insert additional series impedance in your AC supply lines that adversely interacts with filtering designed into your equipment in the first place. Balanced power transformers will cancel ground noise and probably improve the effectiveness of power conditioners or EMI filters. Power regeneration will eliminate noise and power-line harmonics. Shielded power cords will attenuate E-field EMI, and if properly constructed, reduce H-field EMI somewhat, but only between the wall outlet and your equipment.

You have a large choice in how much equipment to add and how much to spend in improving the AC power quality at your audio equipment. In any event, as described in Part 2 of this series, your first step should be to make sure the AC power connection to your audio setup is as good as it can be. ❖

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The Infinite Box Concept, Part 2

This collaboration continues, with listening test results and construction details of open-backed systems. **By G. R. Koonce and R. O. Wright, Jr.**

BREADBOARD LISTENING TEST SETUP

We made the IB breadboard into a full-range system so GRK could listen to a variety of driver/IB combinations. We set a two-way CSB closed-box system (covered in *aX* March and April 2001) on top of the IB breadboard to provide the upper end, and used an electronic crossover (CO) to combine the two boxes. With the CSB woofer about 11″ above the IB woofer, the CO frequency was kept low, at 800Hz. The Ref. #139C driver has a nasty dip about 600Hz, so it used a CO frequency of 400Hz.

Since the CO used is second-order, we hooked the CSB box with inverted polarity, which worked out well. The electronic CO has HP and LP gain controls for matching the relative level of the two boxes. We did all listening with the breadboard system sitting on the floor and several feet from the rear wall.

GRK listened to four IB types with the drivers shown in *Table 9*. The following lists the IB types, which drivers were tested in them, and the available information about expected performance. All IBs have two-2″ damping layers and the DA volume walls lined with nominal 1½″ fiberglass unless noted otherwise:

1)IB type M—Smallest DA volume of any IB tried at 546 in^3. The open rear area was one quarter of the box area, or 22.5 in^2.

  a. Ref. #109K 6.5″ driver—This system should have f_3 around 65Hz with a response peak of about 2dB.

  b. Ref. #111D 6.5″ driver—This driver tested in a slightly smaller DA volume indicated an f_3 around 51Hz with around 2.5dB peaking.

  c. Ref. #119X 8″ driver—This driver tested in a slightly smaller DA volume indicated an f_3 around 62Hz with around 3dB peaking.

2)IB type H—DA volume of 824 in^3, about the correct value for the two 8″ drivers. The open rear area is half the box area, or 45 in^2.

  a. Ref. #119X 8″ driver—This system should have an f_3 around 55Hz and about 3.5dB peaking.

  b. Ref. #139C 8″ driver—This system should have an f_3 around 45Hz and about 3dB peaking.

3)IB type K—DA volume of 1,182 in^3, the maximum capability of the breadboard. The open rear area is one quarter of the box area, or 22.5 in^2.

  a. Ref. #82A 6.5″ driver—This system should have an f_3 around 52Hz with about 3.5dB peaking.

  b. Ref. #111D 6.5″ driver—This system should have an f_3 around 47Hz with about 2dB peaking.

4)IB type H-Mod—This is IB type H (DA = 824 in^3) above with one damping layer removed for a single-2″ damping layer.

  a. Ref. #139C 8″ driver—This system should have an f_3 around 45Hz with greater than 3dB peaking.

BREADBOARD LISTENING RESULTS

We used various CDs to listen to the IB/driver combinations. Quick comparisons were not possible due to the time it takes to reconfigure the breadboard, so the listening sessions stretched over days. When comparing different drivers in the same IB, the front baffle can be changed quickly unless the two drivers

PHOTO 4: Rear view of open box for two-way IB system.

are the same size and use the same baffle. Subjective listening tests lack quantitative results, but we needed to listen to know the IB breadboard "worked," as the testing had indicated.

Ref. #109K in IB type M (DA = 546 in^3)—This had excellent sound, and watching the IB woofer was a joy. On a drum strike it would virtually do a square wave—jumping out, stopping, jumping back, and stopping. The bass was faster and tauter than with a VB using this same woofer built in the past.

The system did not sound bass heavy. Any peaking is low in frequency, helping to avoid that "bass heavy" sound which can occur. We built this woofer into a two-way IB system (discussed later).

Ref. #111D in IB type M (DA = 546 in^3)—GRK could not tell any difference between this and the Ref. #109K woofer, a good-sounding system with similar extension, again very tight bass noted via looking and listening. This would be a system worth building.

Ref. #119X in IB type M (DA = 546 in^3)—We tried the 8″ driver in this small IB and the sound was fine. Again, it had very tight, fast bass. GRK thought the bass extension was not as good as with the 6.5″ drivers, but it is hard to be sure without a direct comparison.

Clearly these results show the IB is rather insensitive to what driver you put into it, as some had predicted. At least this is true with low DA volume IBs. All drivers will sound good in the system, but testing has shown bass extension will suffer if your DA volume is too small.

TABLE 9
DRIVERS USED IN IB BREADBOARD LISTENING SESSIONS

DRIVER	SIZE	f_3	Q_{TS}	V_{AS} — IN^3	NORMAL USAGE
Ref. #82A	6.5″	60.4	1.07	850	Throwaway
Ref. #109K	6.5″	45.6	0.46	1,514	Probably CB
Ref. #111D	6.5″	50.1	0.81	726	CB
Ref. #119X	8″	30.1	0.45	4,804	Probably CB
Ref. #139C	8″	27.6	0.48	3,992	VB till added cone mass

Ref. #119X in IB type H (DA = 824 in^3)—This driver really performed in this slightly larger DA IB. The bass was great and powerful. This woofer would allow building a two-way system. GRK believed this system had better bass extension than this same driver in smaller IB type M, but again it's hard to be sure.

Ref. #139C in IB type H (DA = 824 in^3)—The CO frequency was dropped to 400Hz and GRK could hear the difference as it was turned down from 800Hz. Those nasty woofer dips do affect the sound. This was an impressive system with well-damped bass as indicated both by looking and listening. We built this woofer into a three-way IB (discussed later).

Ref. #111D in IB type K (DA = 1,182 in^3)—While this system sounded good, it did not visually appear to have the good damping of the smaller IBs. For a 6.5″ woofer going down to about 47Hz it was good sound, but GRK thought that the damping was not as good as the earlier systems; it did not have that "IB sound." This was based on both looking and listening.

Ref. #82A in IB type K (DA = 1,182 in^3)—This driver has a Q_{TS} above unity and would thus be a "throwaway." We wanted to see whether the IB held the promise of saving such woofers as testing had indicated. Based on the weakened performance of the Ref. #111D woofer (Q_{TS} = 0.88) in the large DA IB, GRK doubted this woofer had any promise. To his surprise, the Ref. #82A driver was better than the Ref. #111D driver. The damping looked and sounded better and the Ref. #82A would make a good-sounding system.

This does show that the larger DA IBs can still offer good damping and the IB can be used to save a high Q_{TS} driver. The size of an IB does not scale directly with woofer size due to the fixed thickness of the damping layers. For this small woofer the box would be too big, but for a big woofer a large DA IB might be reasonable.

Ref. #139C in IB type H-Mod (DA =

824 in^3)—This is the type H IB with one damping layer removed. As far as GRK could tell, this system sounded the same as with two damping layers. With your hand on the back of the damping layers you could feel the drum strikes far more than with two-2″ layers. We worried that this single 2″-thick layer was not strong enough and decided we will stay with a total of 4″ for woofers. This showed the damping material ought to be mounted in such a way that fore and aft braces retain it near the center as well as at the edges.

SUMMARY OF RESULTS FOR IB BREADBOARD LISTENING

As far as GRK could tell, the IB works as indicated in the testing for DA volumes from 546 to 1,182 in^3. The smaller the design DA volume, the better the approach seems to work. The IB does offer a way to save high Q_{TS} drivers, but with small woofers the box may be physically too big. The Ref. #111D driver seems to work fine in the small IB type M, but did not seem to be as well-damped in the large-DA-volume type K IB.

We have no idea why this one IB/driver combination did not work as well as other combinations. Clearly, there are some things we don't yet understand about the IB approach! Unfortunately, it might mean that for every driver there is some maximum DA volume above which the IB concept does not work; we don't know at this time.

CONSTRUCTION OF IB SYSTEMS

This section discusses the construction and performance of two systems using the IB concept. There is no intent to document the design or detailed construction of these systems. We will show a basic construction approach and discuss results.

Requirements for the boxes were that the damping layers be simple rectangular shapes and that they be removable/changeable. This resulted in a complex construction approach with a removable back; it yields a solid IB system, but simplification would be a valid goal.

First, we constructed a floor-standing two-way system using the Ref. #109 6.5″ woofer and small planar tweeter. This system uses all second-order crossover networks developed via modeling and crosses over at 2.3kHz. We built the boxes, based on the optimum listening angle, with the woofer and tweeter side-by-side (see *SB* 3/97, p. 10) using mirror-imaged boxes so the tweeter was always outboard. The system included a woofer DA volume of 546 in^3 and used a total of 4″ of #705 damping material.

The breadboard tests of IB type M with this driver indicated the system should have f_3 around 65Hz and about 2dB bass peaking toward DSL compensation. Additional bass compensation of about 2dB was designed into the crossover. Testing of one finished box without crossover showed the expected 2dB bass peaking with f_3 a bit higher at 70Hz. Reading f_3 from testing-based plots is always a bit of a guess.

Photo 4 is a rear view of the open cabinet showing the basic construction. The rear of the DA volume is terminated with a "damping panel," which has the largest possible holes while maintaining a ¾″ lip to hold the damping

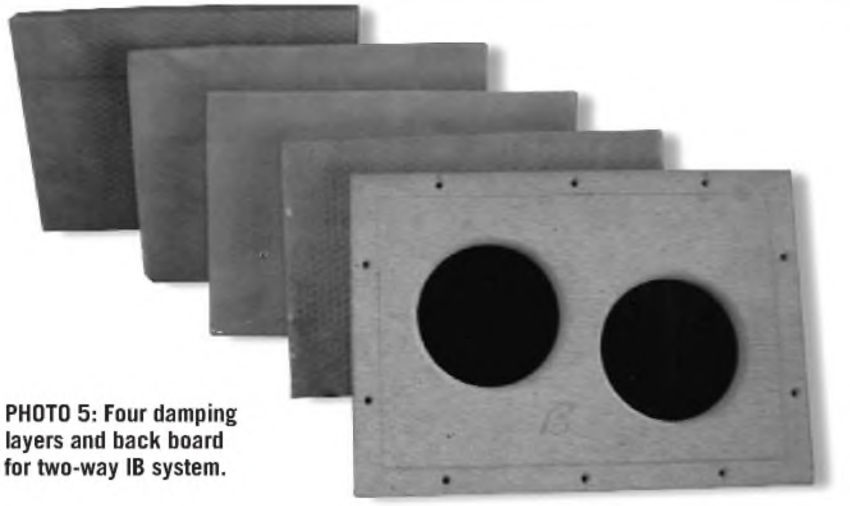

PHOTO 5: Four damping layers and back board for two-way IB system.

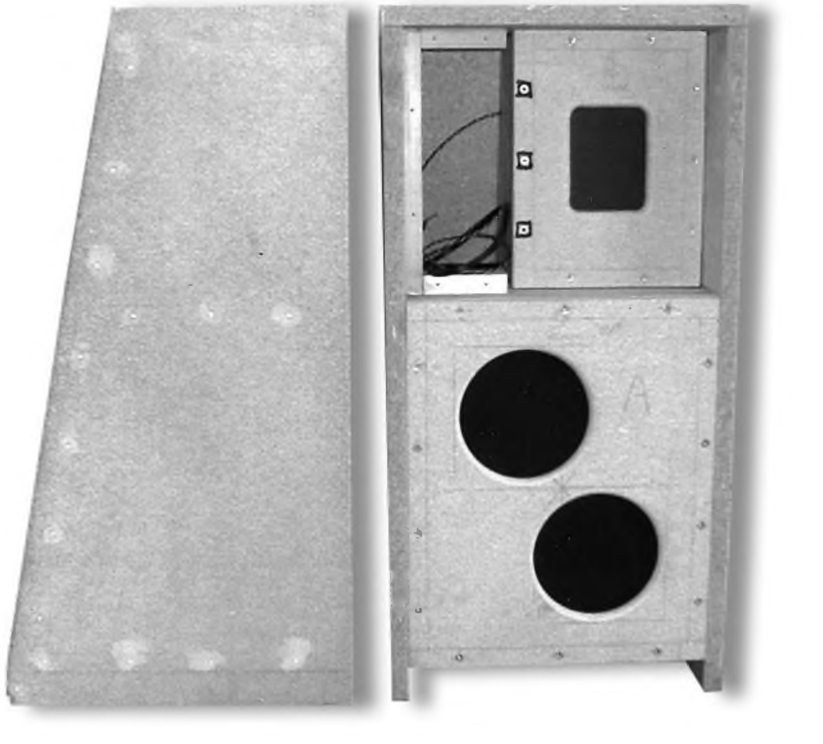

PHOTO 6: Side and rear views of three-way IB system.

layers at their edges augmented by a center rib about 1″ wide. The center rib is strengthened on the driver side, by half of a dowel sliced lengthwise, to prevent it from breaking. The holes in the "damping panel" have a ⅛″ radius routed onto them to prevent cutting the damping material or causing a rattle.

The walls of the DA volume, including the rear of the tweeter box, are covered with nominal ½″ thick fiberglass. Behind the "damping panel" the box is lined on three sides with additional layers of ¾″ particleboard to allow mounting the removable back and providing a constant area "tunnel" for the damping material. The depth from the "damping panel" to the back is 4¼″, but the damping material compresses about ⅛″ when you attach the back. We find the damping material thickness is not exact, so you want to determine this distance for the material you use.

Photo 5 shows the four 1″ thick damping layers (7⅛″ × 11″) and the back of the box. The two holes in the back total just a bit greater (10% to 20% seems good) in area than the driver piston. Again, we cut a ⅛″ radius on both sides of the holes and attached grille cloth to the inside of the back to keep all the damping material, which is fiberglass-based, within the box.

The second system developed is a floor-standing three-way using the Ref. #139 8″ woofer, a 4″ bargain driver as the midrange (Parts Express #299-242), and a Vifa D27TG-15-06 silk dome tweeter. The woofer is an IB using a low dead-air volume of 824 in^3 and 4″ of #705 damping material. The midrange is also an IB with a dead-air volume of about 100 in^3 and 2″ of #705 damping material.

Breadboard testing of IB type I with this woofer indicated a system f_3 of about 45Hz with a peak of about 4.5dB. The fiberglass lining in the DA volume should reduce this peaking to about 3dB for DSL compensation. The crossover is all second-order, with the midrange and tweeter using inverted polarity. The crossover frequencies are 400Hz and 3.2kHz with about another dB of bass boost designed into the crossover.

Basic construction is the same as that used for the two-way system. For both the woofer and midrange the dead-air volume is terminated with a "damping panel." This panel has large holes with a ⅛″ radius and provides a lip, along with a center rib, to retain the damping material on the inboard end. The walls of the woofer DA volume are covered with nominal 1½″ fiberglass, and those of the midrange with nominal ½″ fiberglass.

Behind the "damping panel," ¾″ thick wood is provided on the needed

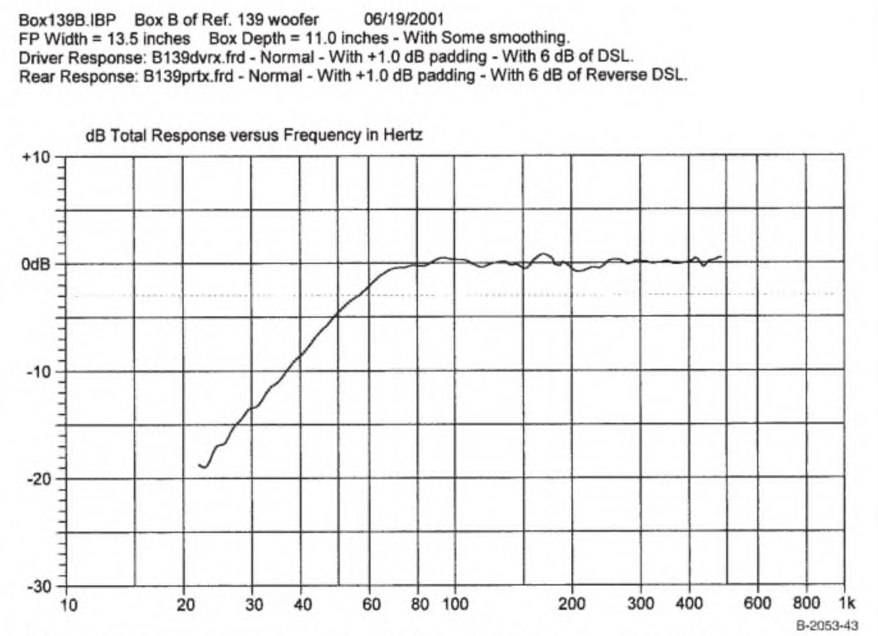

FIGURE 43: Measured response of box with Ref. #139 woofer with 6dB DSL correction.

walls for attachment of the back and a "tunnel" for the damping material. This tunnel is 10½″ × 13″ × 4¼″ deep in the woofer box and 8½″ × 6″ × 1⅛″ deep in the midrange box to match the measured damping material thickness.

Photo 6 shows side and rear views of the three-way systems. You can see that this box is about the same size as would be a three-way 8″ woofer system constructed as a vented or closed box. The woofer back has two holes totaling a bit more in area than the woofer piston, while the midrange uses a single hole again just over the midrange piston area. The holes are cut with a ⅛″ radius and have grille cloth on the inside. The open portion alongside the midrange box is for packing portions of the crossover and thus the brass screws on that side of the midrange back.

LISTENING TO CONSTRUCTED IB SYSTEMS

The two-way systems were finished first and turned out to be delightful. Along with the wider "sweet spot" this configuration gives, the system has very fast and solid bass and "sounds" to have more extension than the measured f_3 around 70Hz would indicate. The surprise was that the midrange was unusually clean, more so than we have come to expect from a two-way system using the woofer up to 2.3kHz. These systems would give many three-ways a run for their money in midrange clarity.

We have no explanation for this other than a possible reduction in Doppler (frequency-modulation) distortion. Whenever a woofer produces the midrange frequencies, you have the problem of the midrange source "moving" at your bass frequencies. This causes Doppler distortion on the midrange sound, which hurts the clarity. Perhaps the high damping on the woofer produced by the IB approach limits extraneous woofer motion and reduces the Doppler distortion that all two-way systems must have.

Initial listening to the three-way systems showed them to sound "bass heavy." Testing of one finished box without crossover showed about 4dB of cone response peaking—more than expected. Also, the leakage was not suppressed as well as in the breadboard. This system has a rather shallow DA volume, and we suspect you are better off building a deep box.

The system response without DSL correction showed about 4dB peaking with f_3 at 43Hz. This system with 6dB DSL correction shows a nearly flat on-axis response (*Fig. 43*). Note that 4dB is about the maximum DSL compensation we recommend for a floor-standing box. The small bass boost in the crossover starts at a higher frequency than the IB peaking, is not needed, and probably accounts for the slight bass-heavy sound.

The small amount of attenuation used on the midrange (0.5–1dB) was removed and the tweeter L-pad corrected; these small adjustments helped remove the bass-heavy sound. This system also has fast, taut bass and is very clean. Bass extension was clearly better than the two-way IB and sounded as extended as the measured f_3 = 43Hz without DSL correction would predict. The midrange sound is very clean and smooth on this system—delightful considering the bargain midrange driver.

IS DSL COMPENSATION WORTHWHILE?

One advantage of the IB technique is that you can build DSL compensation into a system while still maintaining a low system-Q and without going to dual woofers or a dual VC woofer. Is this worthwhile? Most experts recommend making the on-axis response flat, and to do this DSL compensation is needed for any box that does not sit right against the rear wall.

GRK compared the three-way IB boxes with a set of three-way boxes without DSL compensation. Basically listening at stout levels with popular music, he did not hear any great difference. When you turn the music down to moderate or background level, the difference is clear and DSL compensation worthwhile.

Since classical music has true dynamic range, the DSL compensation makes a great difference; the orchestra's tonal balance holds the same "weight" as the playing level changes. Classical piano also showed this same improvement as the playing level changed. Clearly, DSL compensation will allow you to enjoy music played at a lower playing level if you so desire.

Based on these tests and past experience, we still believe the theoretical 6dB of DSL compensation is excessive for a floor-standing box with a low-mounted woofer and recommend about 3–4dB compensation. One of the advantages of the IB approach is that it lets you vary the amount of DSL compensation while many other techniques provide a value of only 6dB. If you desire 6dB of DSL compensation, it is unlikely you can accomplish this by only the IB design approach; a practical limit

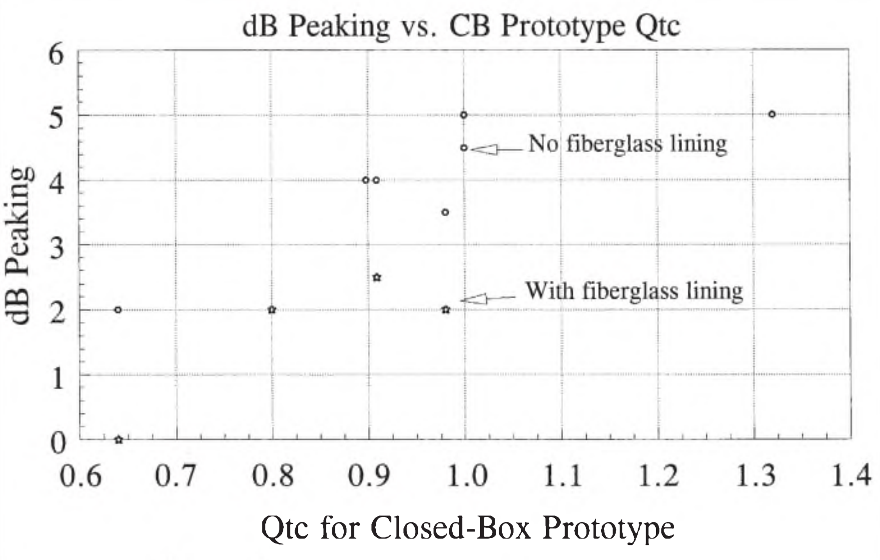

FIGURE 44: Decibel peaking versus CB prototype Q_{TC}.

seems to be in the 3–4dB range.

Reducing the thickness of the damping material or omitting the fiberglass lining in the DA volume may allow more peaking. A shallow DA volume also seems to increase peaking. More systems must be built before we have solid answers on IB peaking.

Including DSL compensation is clearly beneficial for low to moderate listening levels, but keep in mind when you design DSL compensation into an IB the woofer must work rather hard. This combined with the very taut sound of an IB box can lead you to use a playing level that will destroy the woofer, so use some care or a large diameter woofer.

COMPARISON OF OTHER BOX TYPES

It is of interest to see how the IB approach compares to the VB (vented box) and CB (closed box). *Table 10* gives a comparison of the three box types for the drivers offering sufficient IB data. The CB design is for a B2 alignment (Q_{TC} = 0.707) if the driver Q_{TS} permits such a design; otherwise, a C2 alignment at a higher Q_{TC} is shown.

For the VB a BB4 design is shown. This alignment shows a simple peak but results in a much smaller box size for these higher Q_{TS} drivers. We examined a C4 VB alignment, but discarded it because the boxes were huge. If no design is shown it means the driver Q_{TS} does not support a design of that box type.

The peak shown for the IB assumes the DA volume is lined with fiberglass. Keep in mind that the IB will be larger than its DA volume indicates due to the volume added by the damping layers.

In general, the IB has a similar f_3 to the CB designs, but with a smaller box (DA) volume. In many cases, even with the damping layers added, the IB will be physically smaller. The VB designs show a lower f_3, but a much larger box size. Some of the drivers considered have rather high Q_{TS} values for VB application and may result in VBs that are severely power limited. This table does not cover the increased damping of the IB, which produces taut, fast bass the other systems may fail to produce.

HOW TO DESIGN AND BUILD AN IB

The critical design parameter for the IB is the dead-air volume. This is best handled by developing a CB prototype. The number of damping layers is not as critical as you would think and can be played with to trim the system if desired.

1. To design an IB you develop a CB prototype for your driver using design software or equations for a lossless (unfilled) closed box to find a required box volume (V_B). The Q_{TC} used for the CB prototype will establish the shape of the IB response curve and the expected f_3 value. It is impossible for us to predict the exact amount of peaking an IB will have based on the prototype Q_{TC}, because we simply do not have sufficient data for the many variables involved. Clearly, the two important considerations are the prototype Q_{TC} and whether the DA volume has a fiberglass lining.

Figure 44 shows the amount of peaking measured in the IB breadboard tests versus CB prototype Q_{TC} for lined and unlined cases. Clearly, the fiberglass lining reduces the peaking about 1 to 2dB (we have used an average of 1.5dB in this work). If you desire large peaking, then omit the fiberglass lining, keeping in mind that we did not listen to any IB systems using this approach. Going to fewer damping layers will also increase peaking, but at the cost of increased response roughness and possible fracture of the layers. Based on the three-way system built, a shallow DA volume may also increase peaking.

The following are based on 4″ of #705 damping material and a fiberglass lining in a reasonable depth DA volume:

TABLE 10
COMPARISON OF VARIOUS BOX TYPES

DRIVER	BOX TYPE[1]	DA OF V_B[2]	f_3	DB PEAK	Q_{TC}	ALIGNMENT
Ref. #82A	IB	1,182	52	3.5		
	CB					
	VB					
Ref. #109K	IB	546	65	2.0		
	CB	1,123	70	0	0.707	B2
	VB	1,488	45	1.0		BB4
Ref. #111D	IB	1,182	47	2.0		
	CB	1,415	48	1.3	1.0	C2
	VB					
Ref. #119X	IB	1,182	49	2.5		
	CB	3,271	47	0	0.707	B2
	VB	4,444	30	0.8		BB4
Ref. #139C	IB	1,182	40	2.5		
	CB	3,413	47	0	0.707	B2
	VB	4,241	26	1.4		BB4

Notes: 1—IB is an IB design, CB is a closed-box design, VB is a vented-box design.
2—Shows DA or net box volume (V_B) in cubic inches.

a) Designing with a Q_{TC} around 0.6 or below will produce an IB with basically a flat response. The system f_3 will be somewhat below the f_3 value reported for the CB prototype, but we lack data to give a better estimate. This is the approach best-suited to low-Q_{TS} drivers. Keep in mind that very-low-Q_{TS} drivers will yield small boxes with a high f_3 just as they would in any other box type.

b) Designing with Q_{TC} around 0.8 should yield a couple dB of peaking, and again the system f_3 will be somewhat below that of the CB prototype. We used this Q_{TC} value in the two-way system design and it worked out well.

c) Designing with Q_{TC} near unity should yield a peaking of 3-3.5dB and a system f_3 approximately as predicted by the CB prototype. Designing with Q_{TC} around unity is a good approach for drivers with a higher Q_{TS} as would fit a CB design. With very high-Q_{TS} drivers this may produce a physically large box. For low-Q_{TS} drivers this would produce a very small box with high f_3 that might fit a special application. The three-way system we built was based on a CB prototype design with Q_{TC} = 1 and resulted in a good-sounding system with a total box size comparable to that of a VB or CB.

d) If you have a "throwaway" driver with Q_{TS} near unity or slightly above, you can use the IB to "save" this driver. Design it with a Q_{TC} far enough above the driver Q_{TS} to get a reasonably sized box. A design for a driver with Q_{TS} just above unity (1.07) used Q_{TC} = 1.32, and testing showed just 5dB peaking with no fiberglass box lining. This system sounded fine in IB breadboard listening where a fiberglass lining was used. Keep in mind that the physical box size for such designs can be very large.

2. Build the IB with a dead-air volume ahead of the damping material that is about 74% of the prototype VB box volume. The exact DA volume is no more critical than volume is with a CB design, but making it arbitrarily small can produce an IB with a large response peak and high f_3. If you desire more bass peaking, reduce the dead-air volume, remembering that system f_3 will rise. Results with the three-way system we built indicate a deep box may be best.

3. If you keep the front width of the box in the 10″-16″ range, any response peaking will tend to compensate for diffraction spreading loss.

4. A total of 4″ of Owens-Corning #705 damping material will normally be sufficient. Experimenting with as little as 2″ of damping material may produce more bass but with an increased ripple in the response. Up to 6″ of damping material may improve the system response smoothness, especially for IBs with a small DA volume. There are other density Owens-Corning materials available, but we have not evaluated these.

5. You should make the damping material layers nearly the full size of the box. They must be locked into position. On a small breadboard where the damping layers were simply pressed into position, we determined that playing loud music caused the front layer to move up against the driver and the rear layer to move out of the box. We secured the IB breadboard damping layers with silicon rubber, and that worked.

The better approach used in the IB systems constructed is to clamp the damping layers into position about the edges and across the center giving the ability to change the number of layers or replace the layers. The boxes built compressed the layers about ⅛″ when the back was installed, and this worked out well.

6. Cover the walls of the dead-air volume with a fiberglass layer just as you would with any enclosure. This reduces the rear leakage at higher frequencies and helps smooth the system response. It will also help minimize any box resonance. In the boxes built, we used nominal ½″ thick fiberglass for small woofers and midrange drivers, and nominal 1½″ thickness to line the 8″ woofer box. You can try omitting this lining if you desire more peaking.

7. Put a back on the box that limits the open area to a value a bit greater than the cone piston area—about 10% to 20% bigger has worked out well. This helps reduce the rear leakage at high frequency. This rear opening should have a grille cloth on one side to keep the fiberglass in the damping material contained inside the box. The IB breadboard had the "back" at least ⅛″ from the damping material, and such a gap would prevent any possibility of rattles. The systems constructed had the damping layers right against the "damping panel" and back with no problems encountered. We routed the edges of the holes in these panels to a ⅛″ radius—a recommended precaution against cutting the material or having a rattle condition. We did not investigate the effect of a large gap at the back.

8. The box depth affects how the cone and rear-leakage responses sum and can slightly change the system response. The IBs tested ranged from about 11 to 20″ deep. Anything outside this range could produce responses different from those reported.

9. It is possible to implement an IB by making the "open area" on a surface other than the back. We have not tested this and do not recommend it, because the lost RDSL on the leakage will increase the response ripple.

10. It has been "reported" that you can build an IB of thin wood or even cardboard if the walls are covered on the inside with a layer of the damping material. We have not tested this approach but it may work. If you try this approach, we recommend the damping layer be tightly bonded to the walls. Also, that the DA volume be taken as the volume inside the damping-material lining, although we have not yet tested to see whether such a lining raises or lowers the effective box volume. The boxes we constructed went the other way with considerable bracing to assure the front panel and DA volume walls were very stiff.

SAFETY AND OTHER CONSIDERATIONS

Keep in mind that the Owens-Corning #705 damping material is a fiberglass-based product, so you should take the proper precautions when working with it. We have found that the material cuts easily and accurately with a serrated bread knife. We recommend that you cover the rear opening in the box with grille cloth to keep the fiberglass inside the box.

Peaking of the cone response makes the woofer work rather hard. This, combined with the very taut bass of an IB, can lead to quick destruction of a woofer if you are not careful. If you plan to play loudly, you should use a large, high-power woofer with a high-linear-displacement capability.

While the IB breadboard was played quite a bit with the same damping layers, we really have no data on the long-term life of this material in this application. It may be that after a few years of loud playing it fails and must be replaced. Keep this in mind when you construct your box. We also have no data on the long-term effect of this approach on the woofer's life.

All of our testing was done at room temperature. The finished systems have been listened to over a temperature range of about 60–90° with no noticeable change. We have not, however, done any tests of large temperature or humidity changes on this approach. We don't know whether the IB approach is a good technique for application in cars over all weather conditions and other such applications. The Owens-Corning material is used for thermal insulation over a wide temperature range (as in aircraft), so it should not "destruct," but the system response may change.

This article lacks data on IB applications using large drivers with large DA volumes. Subwoofers with 12″ and 15″ high excursion woofers have since been developed to answer questions in these areas. As a follow-up to this article, *aX* will publish an article detailing construction of the 12″ subwoofer.

OVERALL SUMMARY

This study has shown that the IB concept is a valid way to build a speaker system. The dense Owens-Corning #705 damping material can modify the driver's cone response to produce a peaked bass response while maintaining a well-damped system. This material will suppress and phase-shift the driver's rear output to the point that it will slightly aid the bass response rather than oppose it.

The technique thus allows building good-sounding systems that have a bass response peak that can help compensate for diffraction spreading loss. A small portion of this peaking will be obtained by summation of the rear-leakage and cone responses and is thus "free bass"; i.e., it comes without increased cone excursion or additional power into the driver. In practice, probably less than 1dB of free bass is available. The additional bass peaking available via the cone response comes at the cost of additional cone excursion.

The main design consideration in developing an IB is the dead-air volume between the driver and the damping material. If this volume is sized properly, the system response will have a reasonable f_3 value and produce an acceptable shape. Reducing this DA volume by placing the damping material right at the back of the driver reduces the loss through the damping material and produces a system with a high f_3. The IB is rather insensitive to what driver you use, but proper matching of the driver and box can produce a better system.

The main advantage of the IB is the large amount of damping added to the driver on the acoustic side, where it is far more effective. This allows building systems with a response peak while still maintaining a low system total Q. Group delay for the IB system has the shape and magnitude of a closed box using the same driver. Cone excursion would be about the same as for a closed box that matches the IB cone response.

We built one system using the IB concept for the midrange driver with very good results. While not specifically addressed in this work, the Owens-Corning #705 damping material has also been used to cover the inner walls of conventional boxes with good success. It proves to be especially effective in "cleaning up" an open-back midrange driver when used in the midrange chamber.

These things do work; give one a try. Please report your results so a knowledge base of how to apply this technique will develop. ❖

Boston Acoustics Home-Theater Loudspeaker System

Reviewed by Joseph D'Appolito and Ken and Julie Ketler

PHOTO 1: Boston Acoustics Lynnfield VR 6.1-channel surround speaker system. ***(Photos by Ken and Julie Ketler.)***

This is the first time I have tested a home-theater loudspeaker system for *audioXpress*. This test report will be somewhat different from previous test reports since it looks at four different loudspeakers that make up the full system. The system tested comprises the VR-975 left- and right-channel speakers, the VR920 center-channel speaker, the VR-MX surround speakers, and the VR-M/EX rear-channel speaker (*Photo 1*). I will discuss test results for the individual speakers separately and then tie the results together briefly at the end of the report.

THE VR-975 LEFT/RIGHT CHANNEL-SPEAKERS

The VR-975 driver complement (*Photo 2*) consists of a 25mm metal dome tweeter, two 4.5″ midrange drivers, and a built-in 10″ powered subwoofer with a 12″ passive radiator. All drivers are made by Boston Acoustics. The VR-975s are supplied in mirror-image pairs with the subwoofer and passive radiator facing outwards away from the primary listening area. The two 4.5″ midrange drivers are mounted below the tweeter.

ABOUT THE AUTHOR

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

VR-975 IMPEDANCE

The VR-975 impedance is plotted in *Fig. 1*. This is actually just the impedance of the passive portion of the loudspeaker. The rapid impedance rise below 150Hz suggests that this is also the high-pass crossover frequency for the midrange drivers. VR-975 impedance reaches a minimum of just under 4Ω at 150Hz and again at 400Hz. Impedance phase lies between +29° and −14°. This should not be a difficult load for a typical home-theater receiver.

VR-975 FREQUENCY RESPONSE

Because the VR-975 grille cloth is not removable, near-field measurements of the woofer and passive radiator responses were not effective. Instead I made an approximate in-room response measurement using MLSSA's adaptive window function[1]. I placed the speaker on my lab floor so that the nearest vertical reflecting surface was at least 8′ away.

Figure 2 is a plot of the VR-975 on-axis frequency response at two different settings of the subwoofer level control. For this measurement I placed the microphone 1.25m away at tweeter height. In their literature, Boston Acoustics (BA) states that the 9 o'clock position of the control should give the best balance in most rooms. I also measured at the 1 o'clock position. Notice that the woofer level control has almost no effect above 200Hz.

Sensitivity averages 90dB SPL/2.83V/1m between 300Hz and 2kHz, very close to BA's figure of 91dB. There is a broad response dip of 5dB between 2 and 6kHz. At first I thought the response dip was caused by interference between the two midrange drivers. However, moving the mike down to a point midway between the mids produced a more severe dip. You'll see shortly that the probable cause of the dip is edge diffraction. Above 6kHz response rises 5dB to a plateau of 95dB at 10kHz.

Figure 3 is a plot of the VR-975 ⅓ octave pink-noise response. It includes some room effects, but for the most part it is comparable to *Fig. 2*. This response was used to assess low-end in-room extension. Relative to 1kHz the −3dB low-frequency point lies between the 40 and 50Hz ⅓ octave bands. Again, you can see the dip at 4kHz and the peak at 12.5kHz.

VR-975 DRIVER TIMING

VR-975 step response is plotted in *Fig. 4*. The tweeter arrives first, followed in succession by the midranges and the subwoofer. The system is not time-coherent. Notice that the tweeter arrives with positive polarity but the other driver polarities are negative. I'll contrast this with the center speaker response shortly.

VR-975 CUMULATIVE SPECTRAL DECAY

The VR-975 cumulative spectral decay (CSD) response is presented in *Fig. 5*. This waterfall plot shows the frequency content of the system response following a sharp impulsive input at time zero. On the CSD plot, frequency increases from left to right and time moves forward from the rear. Each slice represents a 0.05ms increment of time. The total vertical scale covers a dynamic 35dB range.

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

MANUFACTURER'S SPECIFICATIONS

VR-975

Recommended amplifier power 15–250W
Nominal impedance 8Ω
Frequency response (3dB) 25–20,000Hz
Bass unit 10″ (250mm) DCD polymer-treated
Crossover frequency 150Hz, 2500Hz
Dimensions (H × W × D) 48¼ × 6½ × 15″ (1226 × 165 × 381mm)
Finish charcoal gray cloth with gloss black top
Midrange dual 4½″ (115mm) copolymer
Tweeter 1″ (25mm) anodized aluminum dome
Weight 70 lbs (32kg)
Amplifier power subwoofer—125W continuous
Passive radiator 12″ (305mm) copolymer

VR-920

Recommended amplifier power 15–250W
Nominal impedance 8Ω
Frequency response (3dB) 55–22,000Hz
Bass unit dual 5¼ × 7½″ (135 × 191mm)
Crossover frequency 300, 2700Hz
Dimensions (H × W × D) 6¾ × 27⅛ × 10¼″ (172 × 689 × 261mm)
Finish charcoal gray cloth with gloss black sides and black vinyl veneer
Sensitivity (1W (2.83V) at 1m) 90dB
Midrange 3½″ (89mm)
Tweeter 1″ (25mm)
Weight 32 lbs (15kg)

VR-MX

Recommended amplifier power 15–200W
Nominal impedance 8Ω
Frequency response (3dB) 80–20,000Hz
Bass unit 5¼″ (135mm) copolymer
Crossover frequency 2500Hz
Dimensions (H × W × D) 11 × 11⁵⁄₁₆ × 5-¹⁵⁄₁₆″ (279 × 286 × 151mm)
Finish cherry or black ash wood veneer or white
Tweeter two 1″ (25mm) anodized aluminum dome with AMD
Weight 9 lbs (4.1kg)
Passive radiator 5¼″ (135mm) copolymer

VR-M/EX

Recommended amplifier power 15–200W
Nominal impedance 8Ω
Frequency response (3dB) 110–20,000Hz
Bass unit dual 5¼″ (135mm) copolymer
Crossover frequency 2500Hz
Dimensions (H × W × D) 11 × 11⁵⁄₁₆ × 5-¹⁵⁄₁₆″ (279 × 286 × 151mm)
Finish available in cherry or black ash wood veneer or white
Tweeter dual 1″ (25mm) VR aluminum dome with AMD
Weight 11 lbs (5.0kg)

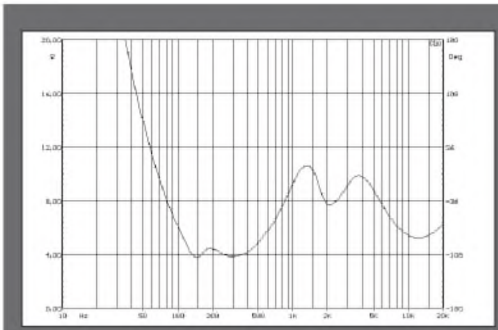

FIGURE 1: VR-975 impedance.

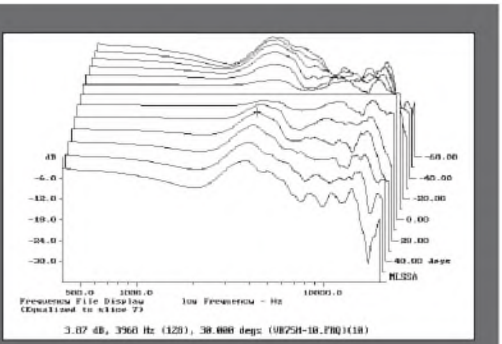

FIGURE 6: VR-975 horizontal polar response.

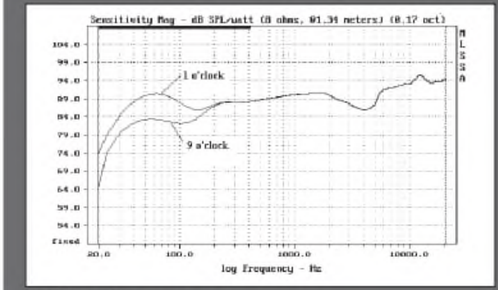

FIGURE 2: VR-975 in-room response.

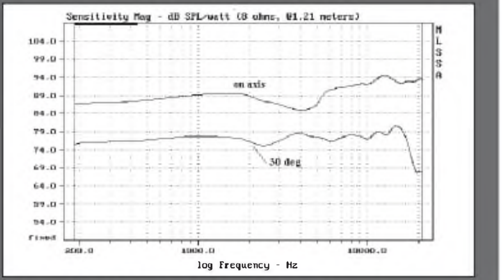

FIGURE 7: VR-975 response at 0 and 30°.

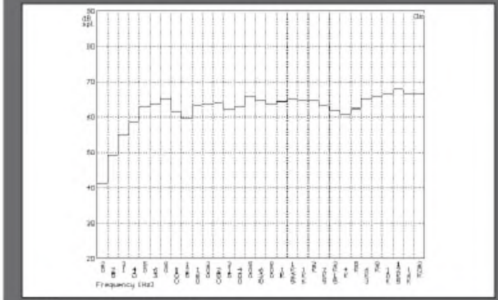

FIGURE 3: VR-975 RTA response.

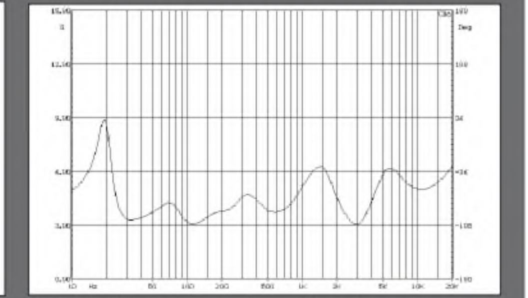

FIGURE 8: VR-920 center-channel impedance.

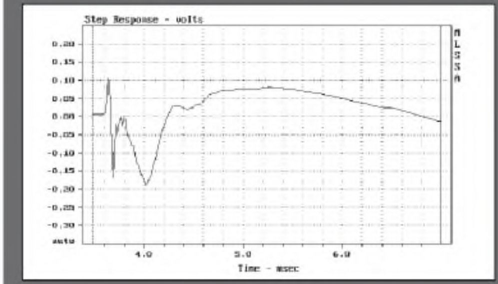

FIGURE 4: VR-975 step response.

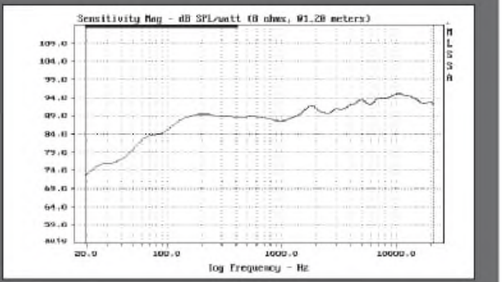

FIGURE 9: VR-920 in-room frequency response.

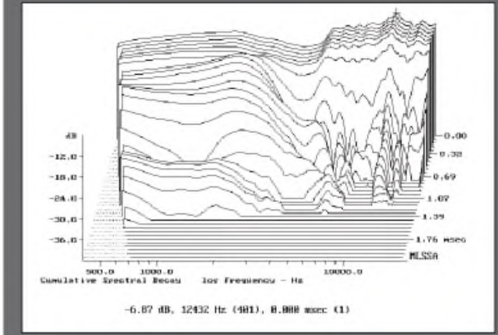

FIGURE 5: VR-975 cumulative spectral decay.

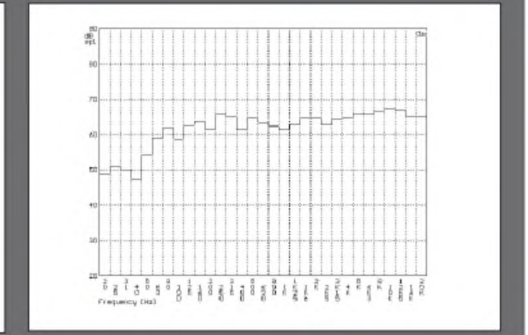

FIGURE 10: VR-920 center-channel RTA response.

Reviewed by Ken and Julie Ketler

If you like teensy-weensy speakers that you can hide within your living room décor, forget the Lynnfield VR 6.1-channel surround speaker system from Boston Acoustics. These speakers are some husky beasts that certainly don't compare size-wise with miniature systems, which have become popular over the last decade. We don't have enough space in this review to answer the age-old "does size really matter?" question. We're not going to touch that one with a ten-foot pole, except to say that it's up to you, especially when talking about speaker systems!

The Lynnfield VR 6.1-channel home theater system includes two front-channel towers (VR-975), a center channel (VR-920), and three surround speakers (two VR-MX and one VR-M/EX). The VR-975 tower weighs in at 70 lbs, holding a tweeter, two midrange drivers, and a built-in powered subwoofer, which includes a 10″ driver, a 12″ passive radiator, and a 125W amplifier. The VR-975s are sleek and attractive with a black "sock" grille and slim footprint. Unfortunately, they seem easy to tip over, especially on a soft, carpeted floor.

Boston Acoustics includes a pair of attachable feet with each tower to help alleviate this problem. Even so, they still aren't completely stable on our floor and have made us nervous for our 2-year-old son, Tommy. They weigh three times as much as he does, for cryin' out loud! We ended up putting a plastic safety gate around the VR-975s as a safety measure. For the remainder of the review, both the baby and the Bostons stayed safely secluded from one another!

The VR-920 center-channel speaker is also big! Luckily, it fits on top of our 36″ TV and out of reach. This enclosure is perfect for anyone with a 36″ (or larger) TV, but could otherwise be a bit on the hefty side. It houses a three-way vented system, which provides a wide and solid center image for dialog and other center-channel information.

The VR-MX left and right surround speakers offer a certain amount of flexibility based on your room arrangement and listening preferences. The enclosures are half-cylinders, which mount flat against your walls. They house two woofer/tweeter pairs, angled away from each other at 90°.

The catch is that one of the woofers is actually a passive radiator. Very slick! The tweeters are apparently wired in phase and operate as quasi-bipoles. Boston Acoustics recommends facing the woofer toward your listening area for "direct" listening (perhaps for music) or away from you for an "ambient" effect (intended for home theater, but up to you, nonetheless).

The third surround speaker, VR-M/EX, is designed for a 6.1 surround system and should be centrally located between the left/right surrounds. It appears very much like the VR-MX surround speakers, except that each "woofer" is actually a woofer and not a passive radiator.

ABOUT THE AUTHORS

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

OUR SETUP

We are using a General Electric DVD player, which is digitally (optically) connected to an Onkyo TX-DS787 A/V receiver. The VR-975s each have a connection for the power cord (included) that feeds the built-in subwoofer amplifier. They have dual binding posts, an RCA jack for line-level inputs and a knob for changing the level of the subwoofer (*Photo 5*).

Setting this control is crucial to the low-end balance of this system. We found that the subs required far less "juice" than the setting of the knob would suggest. This setting took us some time to get right, but sounded superb after we became used to the VR 6.1-channel surround speaker system in our living room—which brings us to another point.

Since we prefer the "natural method" for optimizing our audio systems (i.e., no equalizers), our household has become very comfortable with the idea of having a subwoofer that is separate from the main speakers (that drive everything above 100Hz). As you probably already know, the best spot for the subwoofer isn't necessarily the best spot for the upper-range speakers. Finding the best location for a subwoofer can be very voodoo-esque at times, and large TVs and tower speakers don't fit "just anywhere." So even though speakers have historically contained all drivers in the same enclosure, it seems like a minor step backward to mount them together. That's just our $0.02.

All of the other speakers in this set are very easy to hook up. Each offers a pair of binding posts that accept our 12 AWG speaker wire very nicely. In order to switch the side surround speakers between "ambient" and "direct," it is only necessary to swap the left and right enclosures. As with any home theater setup, you may need to be creative with your wire laying techniques, but it's not a complicated configuration at all.

TEST TRACKS

***Titanic* (DVD)**

JK: Wow! This movie was breathtaking! What a great piece to play on the Boston Acoustics system. I remember watching two distinctly opposite scenes and both were brilliant.

At one point in the story, as *Titanic* is sinking, it is quiet for a minute or two; the only sounds are the creaking of the walls as water trickles into the upper floors. This is eerie and very real.

The opposing scene occurs when the sea is swallowing up the ship and smashing into every wall as the characters Rose and Jack run just inches in front of the rushing water. The sound is loud and clear, making it feel as though we are there on board. Coincidentally, my fingernails are short again!

KK: I think the scene that shows the true capability of the Lynnfield VR 6.1-channel system is when the front of the sinking *Titanic* is already under water, the ship has cracked, and the rear end is pointing straight up in the air, floating momentarily. It's quite spooky to hear people fly past, as they can no longer hold on for their lives. The wreckage creaks around us, reminding us that the horror isn't over.

Suddenly, the remaining section of *Titanic* plunges into the Atlantic, shaking our living room floor, making me forget I'm watching a movie. Very convincing! I later realized that I was so engulfed in this scene that I spilled my drink into my lap. As you might imagine, I thought I had...

***The Sound of Music* (DVD)**

JK: At the opening of the *Sound of Music*, Maria sings in the mountains of Salzburg. The picture and sound transport me there with her—I feel free to run through the grass, too! The Boston Acoustics system wraps the sound around me, coming from the front mostly, but occasionally in the rear speakers, too.

KK: There is one scene that stands out in my mind. When Maria is marrying Georg in the large cathedral, there is a magnificent pipe organ playing their wedding music. The effect of the bass pedals is thunderous. However, one note excites a standing wave mode in our living room that howls annoyingly. All other pedal notes are smooth and deep, but that one...wuuuyoooh! This may be okay for action-thriller movies, which are supposed to be loud and boomy, but for music, yuck. If we were able to reposition the sub(s), perhaps we could have avoided this.

Livingston Taylor "Fly Away" (from *Ink* CD)

JK: There is a wonderful bass part in this piece. It is especially deep, yet the strings are quite clear and very pleasant. Livingston Taylor's voice also comes through very intimately, as though I was listening to him play from the expensive seats in a concert hall.

KK: This "audiophile" acoustic folk recording sounds excellent through the VR-975s. The midrange pluck of the acoustic bass combines seamlessly with the subwoofer's rendition of the body resonance. The acoustic guitar and vocal both sound full-bodied, bright, and shiny without ever becoming sibilant or fatiguing. When putting the receiver in its many synthesized surround modes, all of the other speakers in the Lynnfield VR 6.1-channel surround speaker system perform extremely well and are suitably matched with the front channels.

Chick Corea "Tempus Fugit" (from *Remembering Bud Powell*, 5.1 DTS surround CD)

JK: This piece has a very nice 3D surround sound. The trumpet and saxophones play together tightly,

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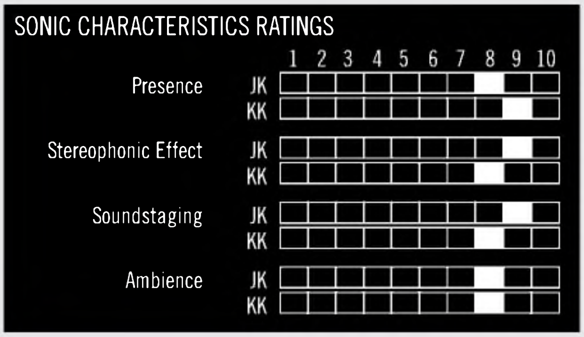

The first time slice in *Fig. 5* (0.00ms) represents the system frequency response. There is a well-organized tweeter mode at 12.4kHz lasting for about 1.25ms. This frequency corresponds to the peak in tweeter response seen in *Fig. 2*. Otherwise, decay response is quite good.

VR-975 HORIZONTAL POLAR RESPONSE

VR-975 horizontal polar response is examined in *Fig. 6*, which is a waterfall plot of horizontal polar response in 10° increments from 60° right (+60°) to 60° left (−60°) when facing the speaker. All off-axis plots are referenced to the on-axis response, which appears as a straight line at 0.00°. Thus, the plotted curves show the *change* in response as you move off-axis. For good stereo imaging the off-axis curves should be smooth replicas of the on-axis response with the possible exception of some tweeter rolloff at higher frequencies and larger off-axis angles. For home-theater applications a more restricted high-frequency response is desirable.

Notice that off-axis response peaks at 4kHz relative to the on-axis position. This means that the on-axis response dip at 4kHz is disappearing as you move off-axis. This is typical of edge-diffraction-induced behavior. You can see this clearly in *Fig. 7*, where on-axis response and response at 30° off-axis are plotted together. The off-axis curve is offset by 10dB for easy comparison against the on-axis curve. You can see that the 4kHz response dip is gone at 30° and overall response is much smoother.

The VR-975 is covered with a grille sock. A raised lip has been placed on the vertical edges of the front baffle to keep the grille cloth from touching the drivers. This lip is causing edge diffraction. For best tonal balance, you should place this speaker facing forward with no toe-in so that the listener is off-axis.

THE VR-920 CENTER CHANNEL SPEAKER

The VR-920 center channel speaker driver complement (*Photo 3*) consists of a 28mm metal dome tweeter and a 3.5″ midrange centrally mounted on the front baffle. A pair of 6″ × 4.5″ oval woofers flanks these drivers.

VR-920 IMPEDANCE

VR-920 impedance is plotted in *Fig. 8*. Although the speaker has two rear-mounted ports, there is no evidence of the typical double-peaked impedance curve of a vented system. (Although not shown, this was confirmed by a second impedance measurement going down to 1Hz.) The vented volume is over damped.

Near-field acoustic measurements show little useful port output. Impedance reaches a minimum of 3Ω at 112Hz and 3kHz. Phase angle lies between +36° and −43°. This may present a problem for some A/V receivers.

VR-920 FREQUENCY RESPONSE

Figure 9 shows the VR-920 frequency response obtained using MLSSA's adaptive window analysis. Best accuracy for this plot is above 200Hz. Response is relatively smooth between 200Hz and 1kHz. Sensitivity in this range is 89dB SPL /2.83V/1m. Response then rises 5dB between 1 and 10kHz and may produce an overly bright-sounding speaker.

Figure 10 is a plot of the VR-920 ⅓ octave pink-noise response, and is fairly representative of the low-frequency extension you will get in a typical room. Relative to 1kHz the −3dB low-frequency point lies between 63 and 50Hz ⅓ octave bands.

VR-920 DRIVER TIMING

VR-920 step response is plotted in *Fig. 11*. The tweeter arrives first at the microphone, followed by the midrange driver and then the

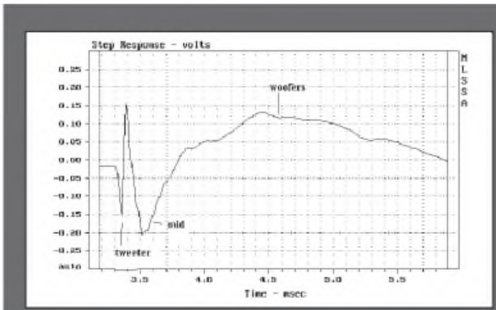

FIGURE 11: VR-920 step response.

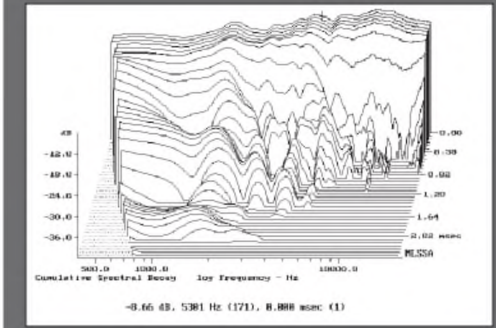

FIGURE 12: VR-920 cumulative spectral decay.

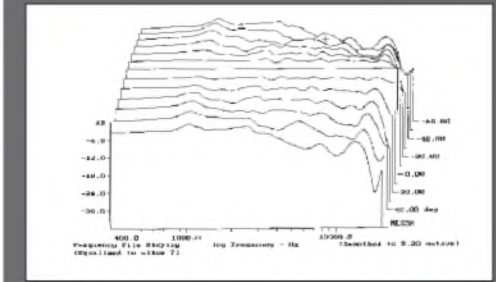

FIGURE 13: VR-920 waterfall plot.

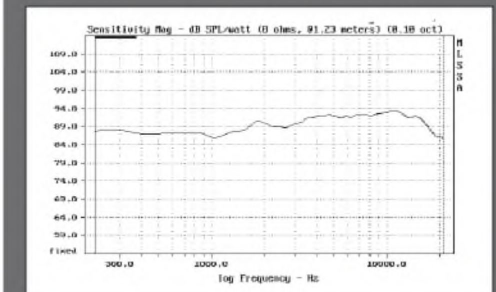

FIGURE 14: VR-920 average response over ±30°.

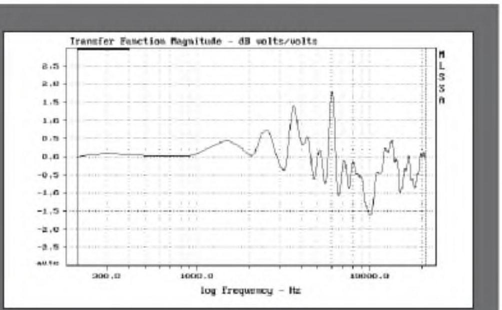

FIGURE 15: Effect of grille on VR-920 response.

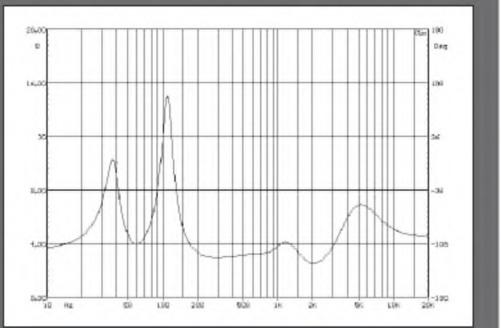

FIGURE 16: VR-MX surround-speaker impedance.

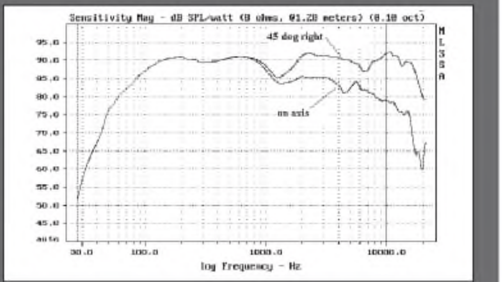

FIGURE 17: VR-MX response on-axis and 45° right.

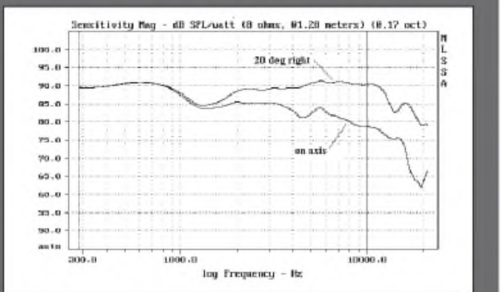

FIGURE 18: Width of diffuse zone.

PHOTO 2: VR-975 main tower speaker (grille removed).

woofers. Notice that the tweeter arrives with negative polarity. This is the opposite of the VR-975 tweeter, which arrives with positive polarity. It will be interesting to see if the reviewers note any audible effect due to this difference. As with the VR-975, the VR-920 does not preserve waveform shape.

VR-920 CUMULATIVE SPECTRAL DECAY

Figure 12 shows the CSD for the VR-920. Tweeter decay time is excellent. The resonant ridge seen in the VR-975 decay is not present here. Although the same tweeter model is used in all BA speakers, the tweeter decay performance here is much better than that of the VR-975. It suggests some sample-to-sample variation in tweeter performance. On the other hand, the VR-920 midrange decay lasts about 0.5ms longer than that of the VR-975.

VR-920 HORIZONTAL POLAR COVERAGE

The center-channel speaker forms the heart of a home-theater system. It defines the focal point for all cinematic action. The center-channel speaker must have uniform horizontal polar response over the viewing region both to preserve the spectral balance of spoken dialog and to center the action for off-axis viewers.

Figure 13 is a waterfall plot of the VR-920 horizontal polar response in 10° increments from 60° right (+60°) to 60° left (−60°) when facing the speaker. With the exception of some high-frequency rolloff, responses at 10°, 20°, and 30° are relatively close to the on-axis response.

Figure 14 is a plot of VR-920 horizontal response above 200Hz averaged over −30° to +30°. Compare this plot with the on-axis response shown in *Fig. 9*. The curves are quite similar. This indicates that there will be little change in spectral balance for off-axis viewers.

THE VR-920 GRILLE

The VR-920 has a removable grille. All test results reported so far have been with the grille off. *Figure 15* shows the VR-920's response with the grille on, but referenced to the response with the grille off. That is, it plots the *change* in response under the two conditions. Below 1kHz the grille has little effect. Above 1kHz, however, the grille causes ragged response deviations of +1.8 to −1.6dB.

THE VR-MX SURROUND SPEAKER

The VR-MX surround speakers (*Photo 4*) are supplied in left/right pairs. The front baffle consists of two panels approximately 90° apart. A 28mm metal-dome tweeter is mounted on each panel. The tweeters are driven with opposite polarity. Mid-bass frequencies are handled by a 5.25″ mid-bass driver mounted on one panel and a 5.25″ passive radiator mounted on the other. As you will shortly see, this configuration approximates dipole response above 2kHz.

VR-MX IMPEDANCE

Figure 16 is a plot of VR-MX impedance magnitude. At low frequencies the plot displays the double-peaked curve of a reflex system. The impedance minimum of 4Ω at 59Hz indicates the box-tuning frequency. There are additional minima of 3Ω at 300Hz and 2.5Ω at 2kHz. Impedance phase lies between +41° and −54° over the full audio range. Again the low impedance points and large phase angles may present a problem to some A/V receivers.

VR-MX FREQUENCY RESPONSE

The VR-MX is designed for wall mounting. The line bisecting the angle between the two front panels at tweeter height defines the on-axis position. The panels are at ±45° with respect to this line. *Figure 17* shows VR-MX frequency response on-axis and at the 45° location. Above 2kHz the on-axis response falls rapidly relative to the 45° position. At 2kHz there is

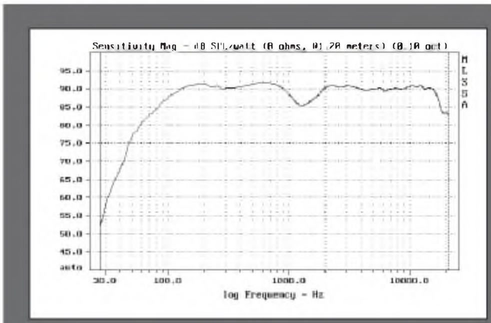

FIGURE 19: VR-MX average response over ±60°.

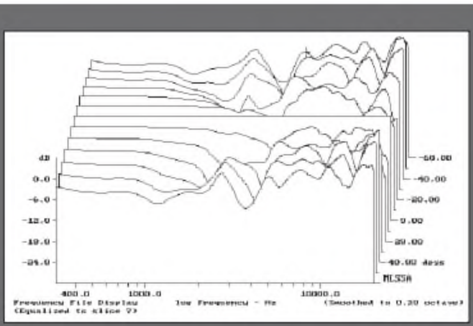

FIGURE 22: VR-MX horizontal polar response.

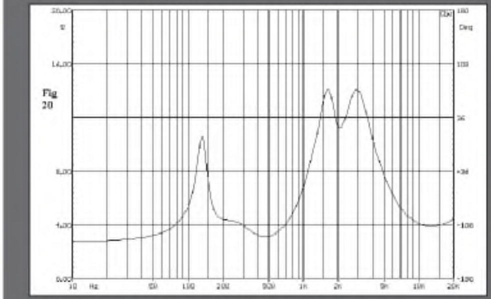

FIGURE 20: VR-MX rear-speaker impedance.

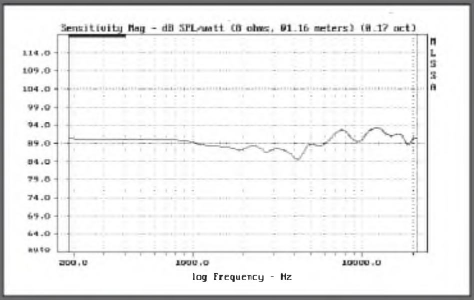

FIGURE 23: VR-MX average horizontal response (±60°).

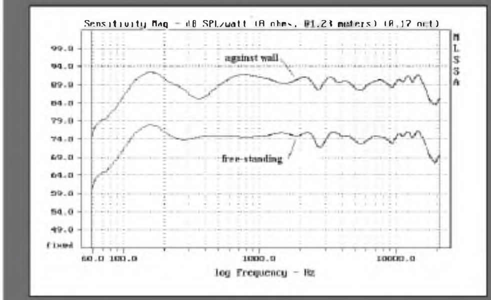

FIGURE 21: VR-MX response against wall and free-standing.

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(from page 48)

forming a wall of horns. The bass sounds so close that I can almost reach out and touch it. "Tempus Fugit" features instruments that cover a wide range, but are each very distinct and clear through the Boston Acoustics speakers.

KK: This is a very fast and furious jazz number. In short, the whole thing sounds great on this system. It's pretty amazing the way the drummer, Roy Haynes, plays very rapid parts (16th and 32nd notes) simultaneously on the ride cymbal and snare drum. The Bostons don't slur a single bounce of the sticks. They each come through very well-focused.

When he runs around the entire drum kit, it's very easy to hear each tom-tom and cymbal. The stereo placement is very well done, due, in some part, to the ambient front-to-back DTS mix. When he finishes a run and hits the crash cymbal and stomps the bass drum, the impact through the Lynnfield VR 6.1-channel surround speaker system is staggering.

Jay Beckenstein "Turnaround" (from *Eye Contact* CD)

JK: During this fun "Turnaround" song, the hi-hat comes in with soft yet clear taps. The bass is very strong and alive without being too rumbly. The sax solo comes in amid the other funky instruments and voices, making the whole section very powerful. The Boston Acoustics system does a great job of conveying this song's power, making us want to get up and dance. Why hold back?

KK: This is a contemporary world-music jazz piece from Spyro Gyra's Jay Beckenstein. It has a nice variety of textures that make this a great test track. There is a lot of percussion including drums, shakers, and chimes that add beautiful shine and movement. The high vocal chants are breathy and clear, while the low parts add a chest-full of power, which support the drums, bass, and keyboards very well.

Beckenstein plays a number of saxophone parts, forming an entire section. Although they meld into one, I can hear each separately if I try to. Bravo, Boston Acoustics!

SUMMARY

When it comes to rating speakers, everyone certainly has his/her own hierarchy of important qualities. In our household, we've pretty much agreed that the overall tone of a speaker stands out as being the most important. Overall, the question is whether we become lost in the movies and music, or do we find ourselves getting hung up on listening to the loudspeakers themselves?

The Boston Acoustics VR 6.1-Channel Surround Speaker System does a great job of carrying us into the world of our recordings. Technically, it is very smooth throughout the midrange, and bright and clear in the treble without ever becoming "spitty." The active subwoofer remains deep and powerful, without ever bottoming out (excuse the pun).

In terms of imaging, it is always easy to close our eyes and "place" the locations of sounds with very little smearing of images. Also, we tend to agree with Boston Acoustics' recommendations for applying the VR-MX surround speakers in direct versus ambient modes.

The Boston Acoustics VR 6.1-Channel Surround Speaker System is very attractive-looking as well, but may rate modestly on the spouse acceptance scale due to its somewhat husky proportions. Overall, a definite contender!

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a 5dB difference, which increases to 13dB at 10kHz. You see that the horizontal response above 2kHz is similar to what you would get from a dipole.

A question of interest: How wide is the dipole region? *Figure 18* shows the response on-axis and at 20° to the right of the on-axis position. Except for some high-frequency tweeter rolloff, the response at 20° matches the response at 45°. This indicates that the dipole region is limited to 20° on either side of the on-axis position.

The perceived spectral balance of the surround speaker is well represented by its average horizontal response. Average horizontal response over ±60° (120°) is shown in *Figure 19*. Except for the 5dB dip at 1400Hz, response is very smooth. Sensitivity averages 90dB SPL/2.83V/1m. Relative to this level the low frequency −3dB point is 94Hz. The 5dB dip appears at all angles.

THE VR-M/EX REAR SPEAKER

The VR-M/EX rear speaker uses the same enclosure as the surround speaker. A 28mm metal-dome tweeter and a 5.25″ woofer are mounted on each panel. The enclosure is sealed, and all drivers are driven in phase.

VR-M/EX IMPEDANCE

Rear speaker impedance is plotted in *Fig. 20*. The single peak of 10.6Ω at 133Hz indicates the speaker resonant frequency. The minimum impedance at 473Hz is 3.1Ω. Phase lies between +53° and −36°.

VR-M/EX FREQUENCY RESPONSE

The rear speaker is designed for wall mounting. I measured on-axis frequency response both against a wall and freestanding. Both responses are shown in *Fig. 21*. The freestanding response is offset by 15dB to ease comparison. Against the wall you see a dip at 360Hz and a broad rise centered on 750Hz. On the other hand, the freestanding response is quite smooth between 300Hz and 2kHz. The dip and rise against the wall are caused by wall reflections. The peak at 160Hz, however, is inherent to the VR-M/EX.

Response ripples above 2kHz are caused by tweeter pair interference. This is seen more clearly in the horizontal polar response waterfall plot of *Fig. 22*. Above 2kHz off-axis response varies erratically with angle. As with the surround speakers, the perceived spectral balance of the rear speaker is well represented by its average horizontal response.

Average horizontal response over ±60° (120°) is shown in *Fig. 23*. Between 200Hz and 1kHz sensitivity averages 90dB SPL/2.83V/1m. There is a broad response dip between 1 and 6kHz. Response then rises above the 90dB level beyond 6kHz.

PHOTO 3: VR-920 center channel speaker (grille removed).

PHOTO 4: VR-M/EX (grille removed) and VR-MX surround speakers.

PHOTO 5: VR-975 input connections and subwoofer control.

SUMMARY

Both the left/right VR-975 and the VR-920 center-channel speakers show an elevated high-frequency response above 4kHz. This may lead to an overly bright sound. This is especially the case with videos that have not been re-equalized for home viewing. The VR-975 can be tamed by listening off-axis. Center-channel horizontal coverage is very good. Off-axis viewers should not suffer any loss of audio quality.

A NOTE ON TESTING

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

Manufacturer's Response:

We wish to thank you for your enthusiastic user review and technical analysis of our VR-975, VR-920, VR-MX, and VR-M/EX home theater speaker package. I would like to take this opportunity to respond to a few criticisms lodged against the VR-975 tower speakers.

First, it would seem from the Ketlers' review that they chose not to use the supplied feet. It is also possible the Ketlers used the feet but not the supplied carpet spikes. We strongly recommend the use of both, especially on thick carpeting. In fact, when the feet are installed, these speakers pass a very stringent "tip test" imposed by the CE for sale of products in the European Economic Community. There are both acoustic and aesthetic benefits to their narrow frontal area. Minimizing the baffle area can enhance imaging, and the speaker's apparent size is minimized, since they are much deeper than they are wide.

The Ketlers refer to a "step backward" as a result of integrating the VR-975's powered subwoofer in the same cabinet as its passive components. They are correct in their assessment of the potential benefits of a separate subwoofer for maximizing performance as a result of room placement flexibility. Whether most consumers take full advantage of this is debatable, since most subwoofers probably end up where they are the least obtrusive, rather than where they are the most effective. Perhaps more importantly, there is the potential to maximize the sonic integration of the subwoofer and the balance of the system by virtue of assembling them in one cabinet. Ideal crossover frequencies and slopes can be chosen, since neither the subwoofer nor the midrange/tweeter are potential variables. A higher, more suitable subwoofer low-pass frequency can be chosen, since there is no concern about the bass localization that can occur in a separate subwoofer. Also, many customers want the bass extension and increased output provided by a powered subwoofer without the need to place an additional speaker component in the room.

Finally, we have one comment about Mr. D'Appolito's evaluation of the VR-MX surround speaker. The VR-MX is not meant to behave as a dipole surround. It uses an active woofer and a passive radiator, in conjunction with two tweeters that are wired out of phase but are driven at different levels. The result is overall higher output on one side of the speaker compared to the other. This allows the VR-MXs, which are sold in mirror-imaged pairs, to be placed beside and behind the listening position and deliver either a more direct or diffused effect, depending upon how they are mounted.

Thank you again for your favorable assessment of our products. ❖

Jeffrey Cowan
Director
Home Audio Products Group
Boston Acoustics

REFERENCE

1. J. D'Appolito, *Testing Loudspeakers*, Audio Amateur Corporation, Peterborough, NH, 1998, Chapter 7.

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Product Review

RCM Akustik DLSA

Reviewed by Richard Honeycutt

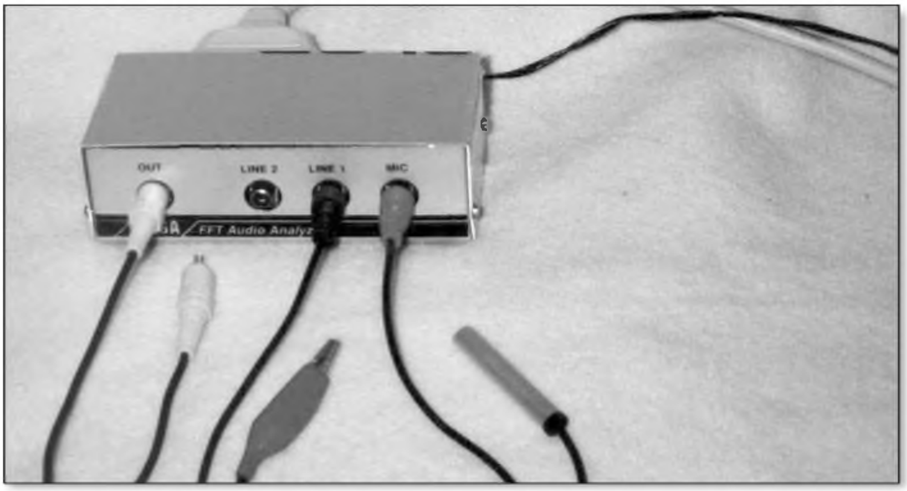

PHOTO 1: The DLSA Pro speaker-test system.

Way back in the bad old days of speaker building, measuring the results of one's efforts was slow, expensive, or both. I remember being in hog heaven when I got my first General Radio chart recorder, logarithmic potentiometer, and tracking oscillator. Of course, by the time the price on even used models had become affordable for me, the unit was out of production, and pens and chart paper were high-priced and hard to find.

Then came the TEF system, followed by SysID, MLSSA, and many others. Much faster, but still pricey. Finally, Bill Waslo introduced the IMP in the pages of *Speaker Builder*, and later, Liberty Audiosuite. Finally, affordable test systems.

SYSTEM FEATURES

Since that time, there has been a slow stream of new introductions in the affordable computerized speaker-test system market. The DLSA Pro, a German entry into the field, consists of a hardware box, a test microphone, a 10:1 attenuator probe, and software that runs on a PC (*Photo 1*). The box contains an A/D converter and interfacing for the mike, probe, and output signals. Available measurements include either impulse-based or MLS-based time, frequency, and phase responses, impedance plots, waterfall plots, and a ⅓-octave real-time analyzer.

The software can convert the time response into an equivalent step response for the unit under test. You can set the FFT record length from 512 to 4096 samples, and sample rate is either 6.4kHz or 51kHz. Hamming, Hanning, and Blackman windows are provided, and you can manually edit the time response to eliminate reflections. You can set synchronous averaging from 1 to 30 measurements. Flat, A, B, and C weighting are available. (For some reason, flat weighting is indicated as "1" weighting.)

The RTA provides a display of present levels as well as either average or peak level (user selectable). The display of amplitude response allows overlaying up to six responses, which can be summed using either magnitude or complex summation. Smoothing of ⅓, ⅙, or 1⁄12 octaves can be applied to the displayed response.

The T/S measurement can use either the added-mass or the test-box method. Provision is made for measuring dB_{SPL} directly, once you have entered the test mike sensitivity, or for inputting the speaker-to-mike distance and reading out the SPL corrected to a distance of 1m. You can also print screen outputs.

The sample unit arrived with a *Quick Installation Guide*, but no manual. The guide contains installation instructions and notes/schematics for the various measurements of which the DLSA Pro is capable. There is no e-mail or website contact info in the manual for dealing with installation or operation issues; instead, you are referred to the ReadMe.txt file on the floppy disk. Telephone, e-mail, and snail-mail addresses for the manufacturer are included in that file.

Also included are instructions about what to do if the computer cannot communicate with the DLSA Pro hardware. Unfortunately, these instructions insist on the use of an "SPP" setting in the CMOS setup of your computer, and such a setting may not be available. (But it may also be unnecessary, as I explain later.) Also, you are warned to remove Epson and/or Canon Bubble Jet printer drivers before trying to use the software.

SETUP AND OPERATION

I installed DLSA Pro on my portable desktop machine without incident. It did not have an "SPP" option in CMOS setup, but the program worked fine anyway, with the machine set on "ECP/EPP." This particular machine does not have any printer drivers loaded; however, if I did have one of the verboten drivers on that machine (Canon and Epson are specifically mentioned), I would not take kindly to having to uninstall it every time I wanted to use the DLSA Pro.

I found operation to be very straightforward and intuitive. Everything is based on menus at the top of the page, and most operation is point-and-click. Once a test procedure is begun, however, it apparently cannot be aborted. Thus, if you begin a Thiele/Small parameter test, for example, but then remember that you have forgotten to input the value of added mass, you must complete the test procedure, then input the correct mass, and go through the procedure again.

On the positive side, procedures such as T/S measurement take you through the process step by step, except for entering the added mass and the reference resistance, which you must remember to do on your own before beginning the procedure. I recommend using the schematics provided in the *Quick Installation Guide*, since some of the screen prompts are not very specific. For example, in the T/S measurement procedure, you are directed to "connect line 1 to test resistor." Of course, a resistor has two terminals, and you are not told which to use. The schematic clears up the confusion.

Measured frequency responses were displayed in an easy-to-read format, and processing time is very short (on a 300MHz computer), so there is almost no delay between clicking the mouse and seeing the response. The RTA is also quite readable, and the dual (present/average or present/peak) display is exceptionally well-thought-out and well-implemented. Not all systems of this type provide A, B, and C weighting, and although their use is more common in other acoustical areas besides audio and speaker testing, they could be handy to have for those other purposes.

The most common use for an RTA in audio work is in setting graphic equalizers in small rooms, and the DLSA Pro would excel in this application, for a fraction of the cost of most other analyzers. This application would be particularly handy in conjunction with a laptop computer.

I performed the T/S test on a sample driver I had. This unit came with the manufacturer's individually-measured T/S parameters. I tested the unit with both the DLSA Pro and the signal generator/voltmeter/frequency counter methods, and the data from both test methods agreed with the manufacturer's data within 5% or so.

The microphone seems to be of the low-priced electret capsule variety, and does not come with individual calibration. This can be a problem if you need accuracy better than ±1dB, or if you'll be measuring frequencies below 50Hz or above 12kHz. Also, unless the capsule has been modified, you cannot use this mike for accurate testing above about $95dB_{SPL}$, because these capsules begin to overdrive the internal FET above that level.

The *Quick Installation Guide* wisely provides a warning against exposing the microphone or other hardware to temperatures above 55°C to avoid possible permanent damage. Also, the 6′ mike cable and 4′ probe and output cables are too short for convenience. Fifteen feet would be much better.

Although I did not use it with a laptop computer, I expect that the DLSA Pro would form the basis of a very handy portable measuring system. Since the unit is very small and lightweight, the 12V DC wall wart could be replaced with a battery pack of your own devising for portable operation. A rechargeable battery pack and a carrying case would be useful accessories, but are not available as of this writing.

SUMMARY

The DLSA Pro seems to be an excellent value for $395. Once you get it running, it does several useful things, and does them well.

With the Liberty Instruments IMP system no longer available, the DLSA Pro is, to my knowledge, the most affordable loudspeaker test system available. For desktop computer applications, I recommend it with only the few qualifications mentioned above. If the manufacturer will include calibration and maximum SPL characteristics on the mike, provide longer cables, solve the issues with printer driver conflicts, and add "abort" buttons on the dialog boxes for test procedures, my recommendation could be unqualified. ❖

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Product Review

The VacuTrace

By Charles Hansen

VacuTrace Vacuum Tube Curve Tracer, Hagerman Technology LLC, PO Box 26437, Honolulu, HI 96825, FAX (808) 394-6076, www.hagtech.com, $999 assembled, $899 kit. One-year warranty.

PHOTO 1: Front panel with 6SN7 plugged into adapter card.

PHOTO 2: Three other adapter cards.

The VacuTrace™ performs measurements of both static and dynamic vacuum-tube characteristics under simulated operating conditions. A plate-supply amplifier and grid-step generator produce voltages that are applied to the device under test (DUT). The effects of these signals on the DUT produce a family of characteristic curves, which are then displayed on an analog oscilloscope (which must have X-Y and preferably Z-axis inputs).

The VacuTrace generates characteristic curves for a number of triode, pentode, and rectifier tubes. It can also display the matching on any two tubes, or the halves of dual triodes. The curves can sweep to 400V plate voltage, –70V grid voltage, 300V screen voltage, with up to 200mA cathode current and 20W plate dissipation. In addition, the VacuTrace provides digital readouts of a number of parameters.

The front panel (*Photo 1*) shows a 6SN7 plugged into one of the four socket adapter cards that connect (one at a time) to the top of the tester, and are held in place with wing nuts. Each adapter card has a rectangular 14-pin male Molex™-style connector that mates with a female connector on top of the VacuTrace. Component pins, which project through the bottom of the adapter cards, are sharp, so you need to handle the board with care.

The rectangular display just below the installed adapter card is the digital readout for the voltages, current, transconductance, and transadmittance. The LED at the left of the display is the status light, which is red in standby mode, green during normal operation, and flashes yellow if there is an overload. The yellow caution LED at the right of the display lights when more than 70V is present at the test connector. The middle and lower rows of the unit are described in *Table 1*.

ADAPTER CARD DESCRIPTIONS

1. Dual triodes adapter, with 9A 9-pin “12AX7/6DJ8” and 8BD octal “6SN7” sockets and a slide switch labeled “6.3V” and “12.6V” to select filament connections. The actual voltage is always 6.3V, so 12.6V tubes such as the 12AX7 have their filaments connected in parallel across 6.3V.

2. Power pentodes adapter, with two 7AC octal “6L6” (6V6, 6L6, KT66, KT88, KT90, 5881, 6550) sockets. There is a wire jumper from pin 1 to 8 on this card that gives it the capability to test the 8EP octal socket 6CA7/EL34, which has a separate suppressor grid connection.

3. Power triode and diode adapter, with a 4D 4-pin socket for “2A3” or “300B,” selected by the left slide switch. The octal 5T “5Y3” (5U4) socket also has a slide switch, labeled “P4” and “P6,” to choose which plate (pin 4 or 6) is operating. I assume you can also test the 5DA socket GZ34/5AR4 tube with its heater-cathode.

4. Blank socket adapter, with circuit board patterns for two octal and two 9-pin miniature sockets. You interconnect the sockets to the 14-pin rectangular connector with jumper wires. The three remaining adapter cards are shown in *Photo 2*.

There are no provisions for 7-pin tubes, given all the variations on pinouts. There is no standard board for 9-pin miniature pentodes, but there is an example on page 15–16 of the manual showing the blank board configured for a 6BQ5/EL84 and 6267/EF86. I did not attempt to customize the blank board, so my 6AV6 and the EL84s that Ed Dell sent me went unused.

PHOTO 3: Rear view of unit.

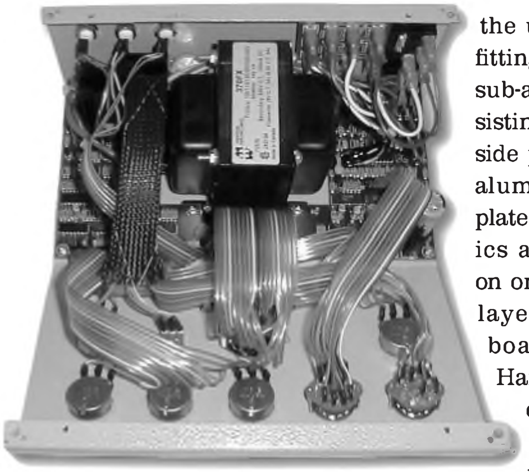


PHOTO 4: Interior view.

The rear panel (*Photo 3*) holds the IEC AC power connector with integral fuse holder, power switch, and three BNC jacks for connecting the VacuTrace output signals to an analog oscilloscope.

A kit version of the VacuTrace is also available. *Photo 4* shows the interior view. The chassis is in two halves, with the upper section fitting into a lower sub-assembly consisting of the wood side panels and an aluminum bottom plate. The electronics are contained on one large multilayer epoxy PC board, with a Hammond 370FX transformer mounted to the chassis. A cutout in the PC board fits around the transformer. All connection to the front and rear panel controls are by means of flat flex cable assemblies.

The component density is quite high on the PC board. I didn't see any surface-mount parts, but the resistors and caps are about as small as they get in through-lead parts. Three MOSFETs and a linear regulator IC are mounted on PC board heatsinks. This is really a professional circuit board design!

TABLE 1

MIDDLE ROW OF CONTROLS (LEFT TO RIGHT)	
Rate/offset control	Sets sweep rate, or grid bias offset voltage in hold mode
Sweep/hold sw	Selects sweep curves, or digital readout in hold mode
Output sw	Selects measurement shown in the digital readout display
Triode/pentode sw	Operates tube as triode (screen tied to plate) or pentode
Screen control	Adjusts screen voltage in pentode mode
LOWER ROW OF CONTROLS (LEFT TO RIGHT)	
Tube select	Selects tube A, B, A/B (overlapped curves), 2A, or standby
Grid steps	Selects step size (gain) for the grid amplifiers (–0.5 to –10V)
Voltage	Sets the maximum value of plate voltage for sweeping
Current	Sets the maximum value of cathode current for a sweep
Power	Sets the maximum plate power dissipation during a sweep

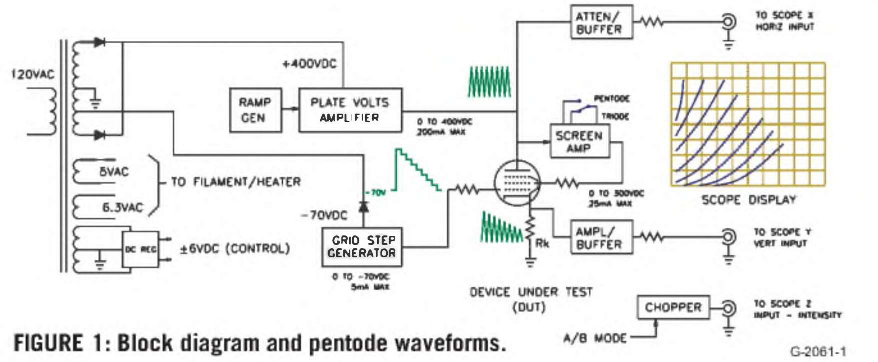


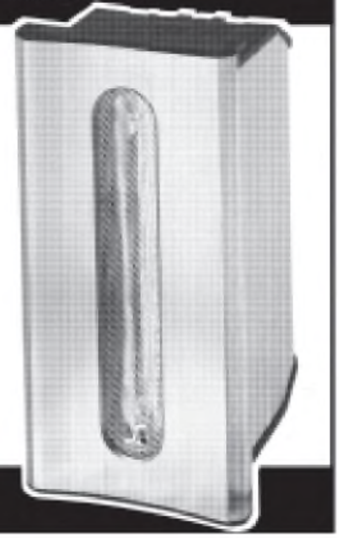
FIGURE 1: Block diagram and pentode waveforms.

The 30-page User's Guide is excellent, with a full nine-page set of schematics. The explanations are brief and well-written, and each function is covered in just the right detail. I was able to test all the vacuum tubes I had on hand after only a brief familiarization with the VacuTrace controls.

FEATURES

I designed a similar Curve Tracer Adapter device for semiconductors[1], and there are some similarities to the VacuTrace. Both require an analog oscilloscope[2], with at least X-Y input capability. The VacuTrace deals with only one polarity for tube tests. A transistor curve tracer's collector/drain supply must deal with NPN/N-channel as well as PNP/P-channel devices.

The VacuTrace grid-step generator needs to generate only negative polarity voltage steps. Bipolar transistors require base current steps of both polarities. FETs require gate voltage steps,



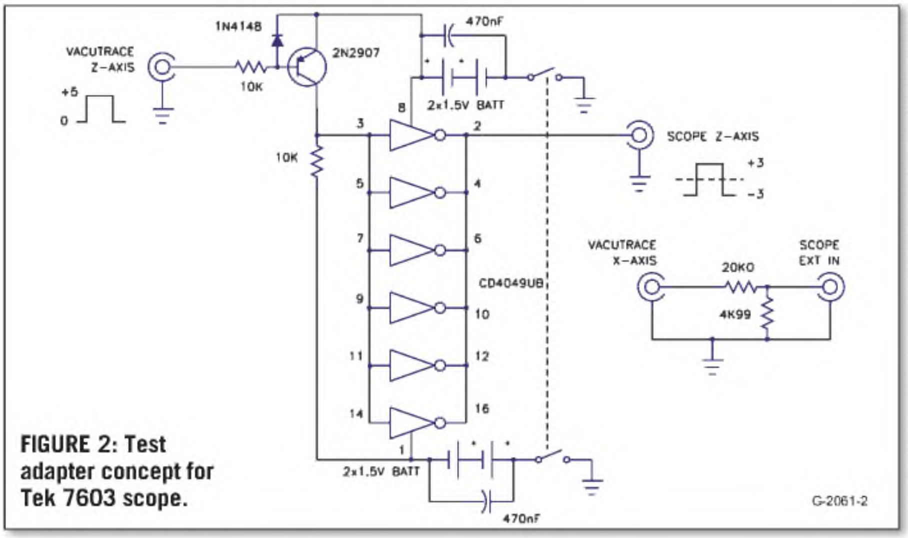


FIGURE 2: Test adapter concept for Tek 7603 scope.

TABLE 2
VACUTRACE TEST DATA FOR SAMPLE TUBES

SAMPLE	GM (µMHO)	µ	RP (Ω)
FSG 6L6-GC #1 (pentode)	5200 (6000)	No spec	43k5 (22k5)
FSG 6L6-GC #9 (pentode)	5450 (6000)	No spec	38k9 (22k5)
FSG 6L6-GC #1 (triode)	4620 (4700)	8.7 (8.0)	1k8 (1k7)
Mullard EL34 #1 matched	12,600 (11,000)	No spec	23k8 (15k)
Mullard EL34 #2 matched	12,600 (11,000)	No spec	22k2 (15k)
GE 6SN7GTB "A"	2220 (2600)	21.9 (20)	9k9 (7k7)
GE 6SN7GTB "B"	2230 (2600)	21.7 (20)	9k7 (7k7)
Philips 5814A/12AU7 "A"	2500 (2200)	21.0 (17)	8k4 (7k7)
Philips 5814A/12AU7 "B"	2530 (2200)	20.7 (17)	8k2 (7k7)
Sovtek 12AX7WA "A"	1580 (1600)	122 (100)	76k9 (62k5)
Sovtek 12AX7WA "B"	1670 (1600)	111 (100)	66k7 (62k5)

and the polarity differs not only with N or P types, but also with whether the device under test has a depletion mode or enhancement mode gate. A transistor curve tracer also supports testing of thyristors, IGBTs, unijunction transistors, and (without use of the base/gate generator) diodes and zeners.

This is not to portray the VacuTrace as a bare-bones test tool. It has the very useful digital readout, lots of internal protection (for the tester—if you intentionally overload your tubes, that is your choice), plate and grid supplies for two tubes, and a Z-axis signal that lets you see the curves for these two tubes simultaneously. If your scope has a Z-axis (intensity) input, a 48kHz chopper modulates the B tube trace so it appears as a dotted line.

Also, my curve tracer just full-wave-rectified the 60Hz collector/drain-supply transformer to simplify the sweep supply design. The VacuTrace uses power MOSFETs driven by a ramp generator to produce a higher-frequency triangle-shaped plate-voltage sweep for higher persistence and better definition at the sweep peaks.

OPERATION

Figure 1 shows a block diagram of the VacuTrace testing a triode-connected pentode. Voltages corresponding to the combined effects of the triangle plate-voltage supply, the grid-step voltage generator, and the voltage across cathode-current sense resistor Rk are applied through op-amp attenuator/buffers to the oscilloscope X-axis (horizontal input), Y-axis (vertical input), and Z-axis via three BNC jacks located on the rear panel. The scope channels must be DC-coupled, but you don't need a wide bandwidth on any of the channels.

If using a single-channel scope, turn the time-base switch to the X or external position. A two-channel scope with an XY time-base switch position can use one channel as the X channel, and the other channel as the Y channel. Appropriate scaling factors (volts/current per division) are given in the User's Manual.

There are two ways to display curves: cathode current versus plate voltage (*Fig. 1*) and cathode current versus grid voltage. The waveforms for the plate-voltage amplifier, the grid-step voltage, and the resultant plate sweep curve scope display (Vp versus Ik) are also shown in *Fig. 1*. Note that the screen is driven from a separate MOSFET amplifier, which in pentode mode derives its adjustable supply from the plate-voltage sweep. In triode mode, the screen grid is connected to the plate supply through the limiting resistor.

There are two filament voltage windings available: 5V AC and 6.3V AC. As mentioned previously, tubes such as the 12AX7 have their filaments connected in parallel across the 6.3V winding. A separate power supply is used to provide ±6V DC for the op amps and other control circuits. The digital logic is CMOS, and it uses the +6V DC side of this bipolar supply.

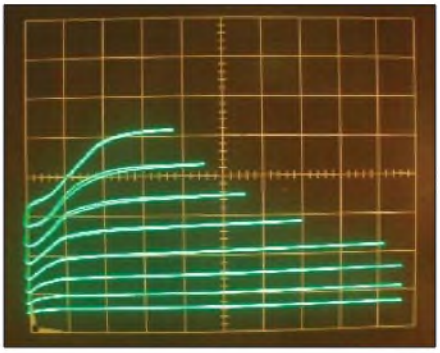


FIGURE 3: VacuTrace curves for 6L6 "A" (Sample #1).

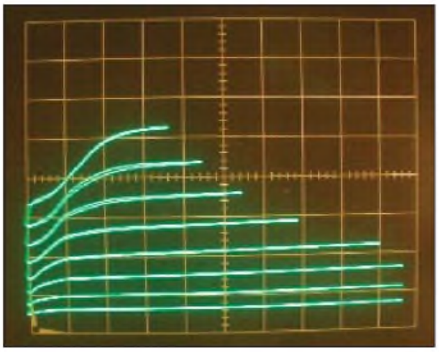


FIGURE 4: VacuTrace curves for 6L6 "B" (Sample #9).

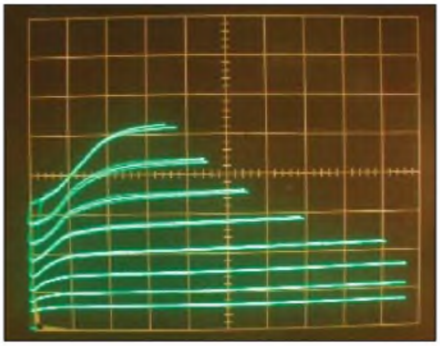


FIGURE 5: VacuTrace "A/B" curve for two 6L6 samples.

The grid-step generator provides grid voltage, stepped in eight equal increments by a staircase, or step generator, on successive sweeps of the plate voltage. The first step has a value of zero. It then decreases at a rate of one step per cycle of the plate sweep (value selected by the grid steps switch) down to seven times that value, and then returns to zero after the eight steps are complete. The pattern repeats as long as the VacuTrace is in the sweep mode.

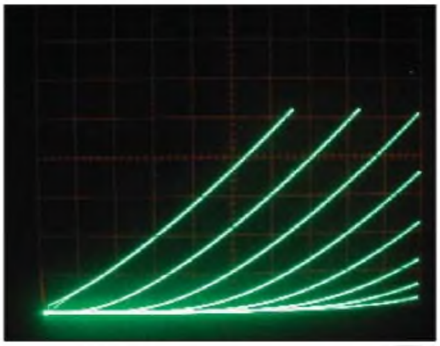

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FIGURE 6: VacuTrace curves for 6L6 sample #1, triode mode.


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FIGURE 7: VacuTrace curves for matched EL34 "A" (sample #1).


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FIGURE 8: VacuTrace curves for matched EL34 "B" (sample #2).

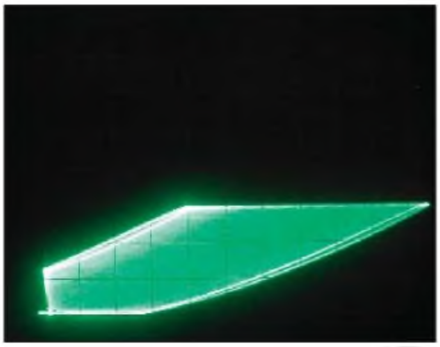
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FIGURE 9: VacuTrace 5Y3 pin-6 plate curve, current limiting.

Cathode current through the DUT flows through the cathode resistor, and the voltage drop across it provides the vertical (Y-axis) signal. The sensitivity of the plate and grid voltages and cathode current is set by their respective controls. Note that the 20Ω cathode-current sense resistor causes a bit of degeneration that introduces small errors in the sweep curves, since the actual grid-cathode voltage changes as a function of cathode current. VacuTrace subtracts out this error for Vg and gm measurements, but the error remains in the scope-displayed traces.

The cure for this would be another transformer, so the grid voltage could be floated between the grid and cathode. This would be a nonstandard transformer voltage; both expensive and space-consuming. (This extra supply is absolutely necessary for the base/gate supply in a transistor curve tracer, be-

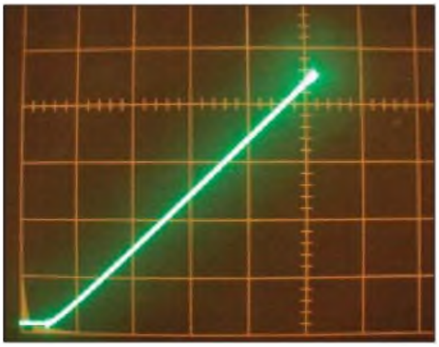
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FIGURE 10: VacuTrace 5Y3 pin-6 plate curve, 2A mode.

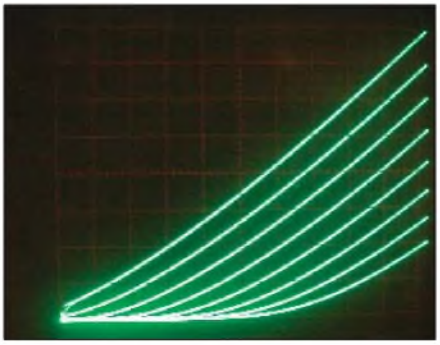
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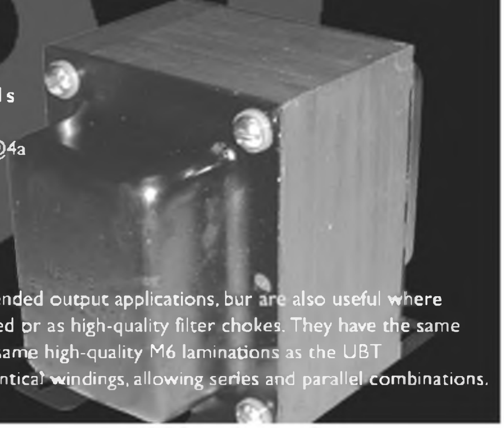
FIGURE 11: VacuTrace 6SN7 curves.



cause the base current is a significant portion of the emitter current. In order to convey actual collector current, it must not flow through the emitter current-sensing resistor.)

In the hold mode, the sweep is disabled, and the Vp plate voltage remains at the preset limit setting displayed on the scope X-axis in sweep mode. The other fixed voltages and currents (Vs, Vg, Ik) are easily measured by means of the VacuTrace's LED digital display. Grid voltage appears as a positive number in the display ("20" means –20Vg), and transconductance is displayed in mmhos, not the more common μmhos given in the tube manuals.

In transconductance (gm) mode, a 625Hz modulation is added to the grid voltage. The dynamic peak-peak grid voltage and resulting AC component of the cathode-current modulation is used to calculate transconductance. Similarly, in transadmittance (gp) mode, the plate voltage is modulated and the dynamic cathode current is used to determine output conductance.

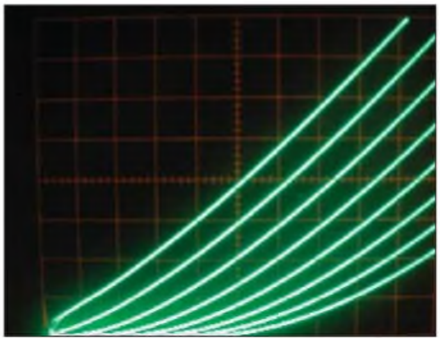


FIGURE 12: VacuTrace 12AU7 curves.

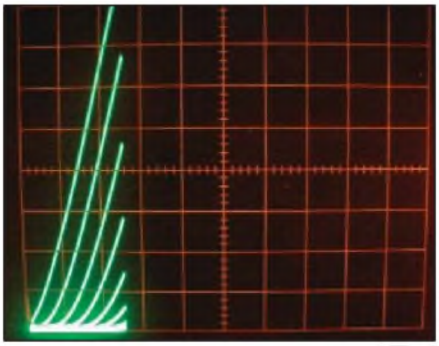


FIGURE 13: VacuTrace 12AX7 curves.

TEST DATA

I had a number of tubes on hand to test the VacuTrace:

FSG 6L6-GC (two)
IEC/Mullard 6CA7/EL34 laboratory balanced matched pair
Sovtek 5Y3GT
G.E. NOS 6SN7GTB
Philips ECG JAN 5814A/12AU7
Sovtek 7025/12AX7WA

I hooked up my analog scope (a Tek

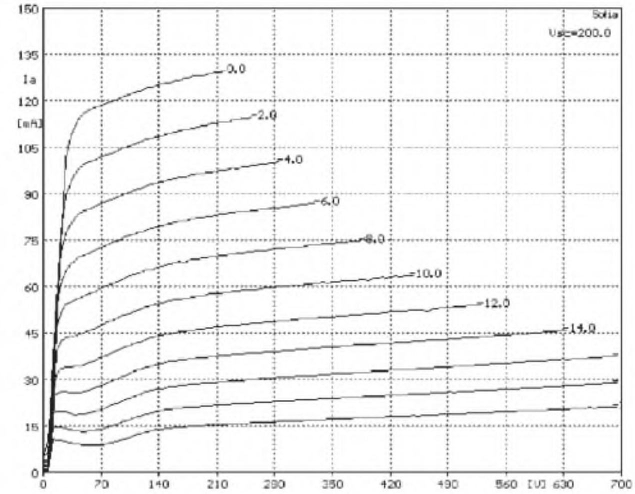


FIGURE 14: Sofia curves for 6L6 "A" (Sample #1).

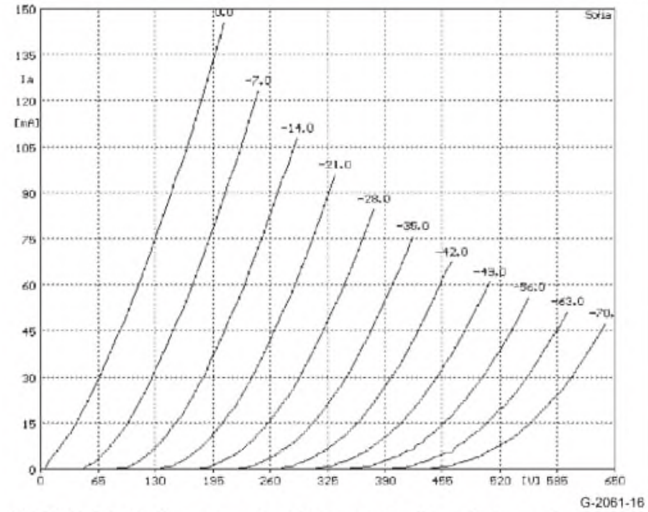


FIGURE 16: Sofia curves for 6L6 sample #1, triode mode.

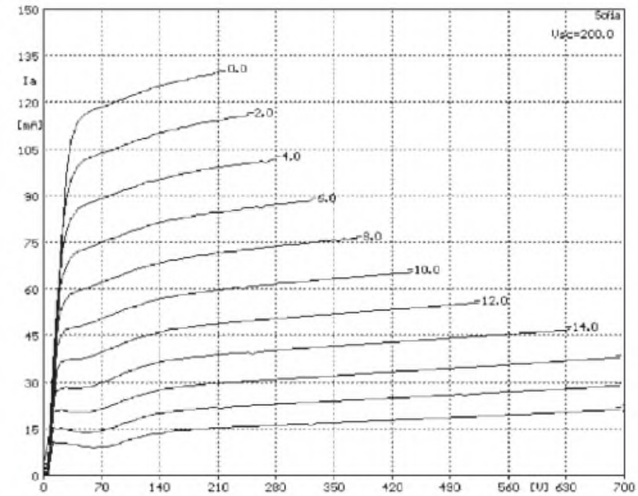


FIGURE 15: Sofia curves for 6L6 "B" (Sample #9).

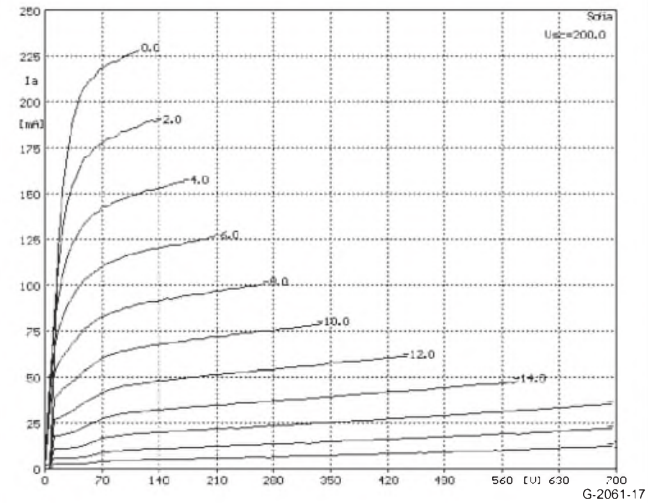


FIGURE 17: Sofia curves for matched EL34 "A" (sample #1).

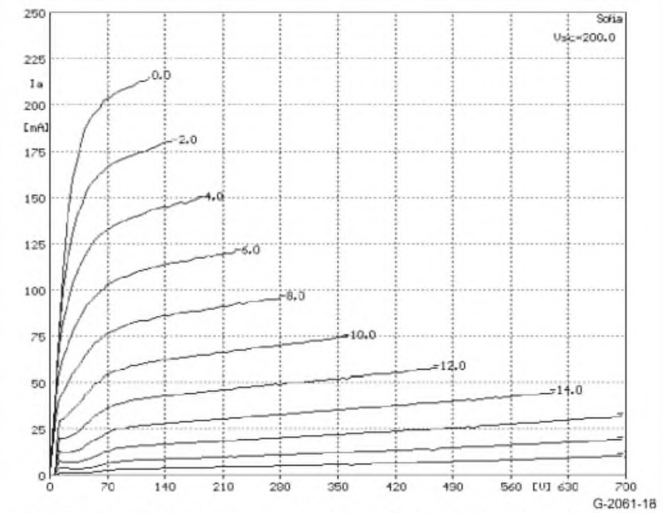


FIGURE 18: Sofia curves for matched EL34 "B" (sample #2).

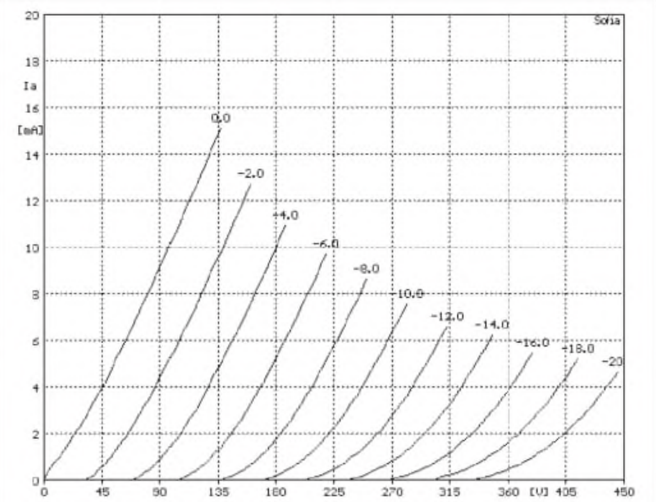


FIGURE 19: Sofia 6SN7 section "A" curve.

7603) to the three VacuTrace output BNC connectors. First, I installed the two 6L6 tubes, set up the controls and tested the "A" tube of the pair. It took some fiddling with the controls to make the most of the available graticule area. My first problem was that my X-axis input has only three ranges: EXT with ×10 Magnifier, EXT, and EXT÷10. This is equivalent to 10mV/div, 0.1V/div, and 1V/div, respectively.

The first two ranges drive the horizontal (plate voltage) traces off-screen too soon, while the EXT÷10 range makes the curves very small. The 7B53 is the best time-base plug-in Tek made for this scope, so I can't get any finer resolution. Some of the newer two-channel scopes (20MHz Tenma, B&K, Hameg, Protek, Hitachi, and so on) can run XY mode directly on the X input. This gives you an attenuator with lots of 1-2-5 decade settings.

Problem number two was that the VacuTrace TTL Z-axis output did not blank the "B" tube curve in "A/B" matching mode. A positive voltage will blank the 7603 scope, but its Z-axis input impedance is only 500Ω. This formed a voltage divider with the VacuTrace Z-axis open collector output resistor that left too little voltage.

I designed (but did not build or test) a test adapter for the Tek 7603 based on the input/output characteristics of the VacuTrace and scope (*Fig. 2*). The Z-axis circuit on the left translates the 48kHz TTL-level signal to ±3V DC using a transistor and a CD4049B hex inverting buffer. Four AA cells power the circuit.

The passive EXT input adapter on the right uses a 5:1 voltage divider that is compatible with the 100Ω VacuTrace V_{plate} buffer output and the 1M input impedance of the EXT amp input to the scope. You should be able to adapt these interface concepts to any scope as long as you have the scope specs and schematics.

For the scope photos in this article (*Figs. 3–13*), I settled on the EXT input and limited the voltage, current, and power settings so the curves would fill the screen. When I took measured data with the LED display, I used the EXT÷10 range to set the plate voltage before making the measurements. My 6L6 scope settings were as follows:

Y-input = 0.5V/div (12.5mA/div at the tube)

X-input = EXT (equivalent to 10V/div at the tube)

For the other tubes, I adjusted the grid step, voltages, current, and power settings for the best display. The triodes needed more vertical sensitivity for good sweep traces.

I also used the VacuTrace's LED display to measure some of the tube parameters, which are listed in *Table 2*. I used the typical operating points for Vp = 250V DC in the *Sylvania Receiving Tubes Technical Manual*. The measured data is followed by the published data in parentheses. I have converted the VacuTrace transconductance readings to µmhos.

The curves for 6L6 "A" (sample #1) are shown in *Fig. 3*. The curves for 6L6 "B" (sample #9) are shown in *Fig. 4*. As you can see, there is some "looping" during the retrace with decreasing B+ voltage. The looping is caused by two things.

In triode mode it is due to the bandwidth limitation of the grid amplifier. You can compensate for this by slowing down the repetition rate. The lower the peak plate voltage, the shorter the sweep and hence the faster the grid amplifier must recover. The grid amplifier is tuned for a step response and is rather insensitive to load capacitance.

The largest loop is seen when the grid must step from most negative to zero. This is where you see a small loop near zero plate voltage. As the plate ramp sweeps back down, the grid has settled and the curve is correct.

With pentodes there is an additional

TABLE 3
SOFIA TEST DATA FOR SAMPLE TUBES

SAMPLE	GM (µMHO)	µ	RP (Ω)
FSG 6L6-GC #1 (pentode)	6830 (6000)	246 (N/S)	36k1 (22k5)
FSG 6L6-GC #9 (pentode)	6610 (6000)	214 (N/S)	32k4 (22k5)
FSG 6L6-GC #1 (triode)	5750 (4700)	9.7 (8.0)	1k7 (1k7)
FSG 6L6-GC #9 (triode)	5590 (4700)	9.5 (8.0)	1k7 (1k7)
Mullard EL34 #1 (pentode)	19,800 (11,000)	316 (N/S)	15k9 (15k)
Mullard EL34 #2 (pentode)	14,700 (11,000)	228 (N/S)	15k4 (15k)
Mullard EL34 #3 (pentode)	15,200 (11,000)	387 (N/S)	25k5 (15k)
Mullard EL34 #4 (pentode)	14,700 (11,000)	321 (N/S)	21k8 (15k)
GE 6SN7GTB "A"	2390 (2600)	23.8 (20)	8k8 (7k7)
GE 6SN7GTB "B"	2860 (2600)	24.2 (20)	8k3 (7k7)
Philips 5814A/12AU7 "A"	2500 (2200)	19.5 (17)	7k8 (7k7)
Philips 5814A/12AU7 "B"	2500 (2200)	19.9 (17)	7k9 (7k7)
Sovtek 12AX7WA "A"	1500 (1600)	101 (100)	67k1 (62k5)
Sovtek 12AX7WA "B"	1660 (1600)	95.8 (100)	57k6 (62k5)

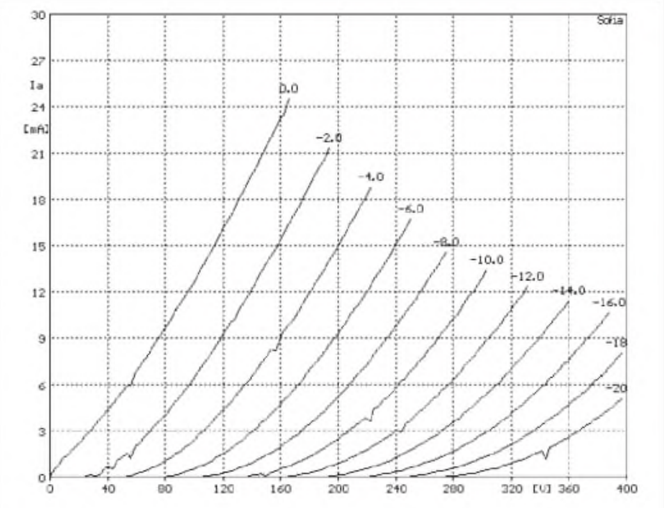


FIGURE 20: Sofia 12AU7 section "A" curve. G-2061-20

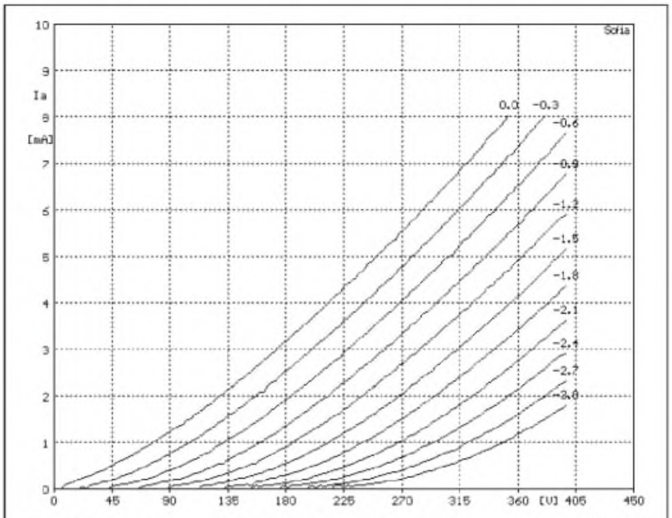


FIGURE 21: Sofia 12AX7 section "A" curve. G-2061-21

problem: the current drawn from the screen supply. The VacuTrace screen amplifier does not have a real low impedance and will droop a bit under heavy load. This causes pentode curves to split. Unlike the other looping problem, running a faster sweep rate will help cure this one.

Tube element capacitance and stray capacitance in the test fixture also allow some of the plate current to bypass the tube itself. Laboratory analog semiconductor curve tracers (such as the Tek 576 and 577) use elaborate capacitance cancellation circuitry that is beyond the scope (no pun intended) of the VacuTrace design, since each tube under test will have a different load. This is not a major problem. After minimizing looping as just described, the best approximation of the actual characteristic curve when looping is present is an imaginary straight line down the middle of the loop.

I have included the "A/B" trace for the two tubes in *Fig. 5*, even though the "B" trace is not blanked (see fix in *Fig. 2* for the Tek 7603 scope). You can still differentiate between the two tube curves for matching purposes, so the VacuTrace is useful even for scopes without a Z-axis input capability. The VacuTrace shares a common plate supply when two tubes are used for comparison. It alternates between the two tubes by cycling each grid into cutoff.

Since there were good specifications for the triode-connected 6L6 in the tube manual, I ran sample #1 in triode mode. The curves are shown in *Fig. 6*. Next up were the IEC Mullard EL34s, which came in a single box marked "lab-matched balanced pair." According to the scope traces (*Figs. 7* and *8*) for these two tubes, they weren't matched very well at all.

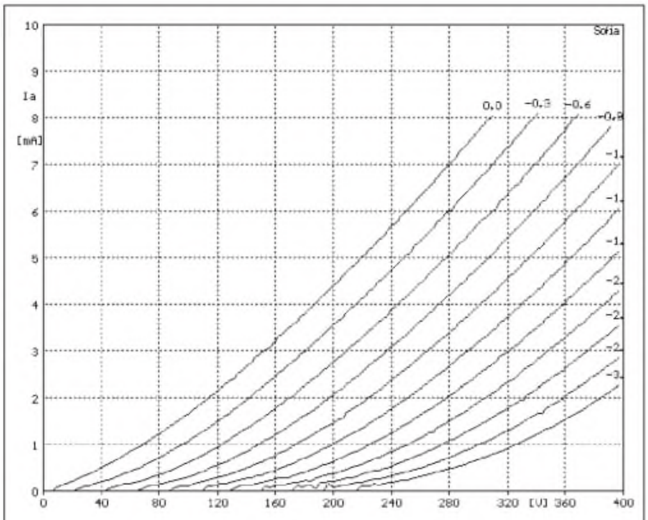


FIGURE 22: Sofia 12AX7 section "B" curve. G-2061-22

The 5Y3 curves do not use the grid step input. *Figure 9* is the pin-6 plate curve in "A" mode, which shows a blur of curves. Jim Hagerman explained that tracing diodes is more difficult because they tend to trigger the shutdown circuit so fast. They draw high current at low plate voltages, while the VacuTrace is best at delivering high current at high voltages (or low power dissipation on the plate supply), making it the most accurate with triodes.

As he suggested, things were better with the 5Y3 in "2A" mode, which doubles the current and power, providing plate current up to 200mA. *Figure 10* shows the single 5Y3 trace with the voltage and current reduced, in "2A" mode.

The dual triode adapter has one octal and one 9-pin socket. You can use only one socket at a time. Selecting "A" or "B" switches between the two sections of the triode, while "A/B" shows both curves simultaneously.

The "A" section curves for the 6SN7 are shown in *Fig. 11*. The "B" curves were so close that I couldn't see any appreciable difference in "A/B" mode. This close match is confirmed by the measured data for the two sections in *Table 2*.

My next triode test was with the JAN 5814A/12AU7 (*Fig. 12*). Again, the curves for the two sections were nearly identical. And look how similar they are to the 6SN7 in *Fig. 11*.

Figure 13 gives the curves for the "A" section of the 12AX7, whose "B" section curves did show a noticeable difference (not shown). This is reflected in the wider spread of measured data in *Table 2*. I had to use the EXT÷10 scope setting here to get the best display. The 0–250V DC plate sweep is now crammed into the first quarter of the horizontal display.

The plate resistances, rp, for the 6L6s in pentode mode (calculated from 1/gp) are about 50% high, while in triode

TABLE 4
SOFIA ERR MATCHING DATA FOR SAMPLE TUBES

SAMPLE	SAMPLE	ERR
FSG 6L6-GC #1 (pentode)	FSG 6L6-GC #9 (pentode)	110
FSG 6L6-GC #1 (triode)	FSG 6L6-GC #9 (triode)	48
Mullard EL34 #1 (pentode)	Mullard EL34 #2 (pentode)	245
Mullard EL34 #3 (pentode)	Mullard EL34 #4 (pentode)	192
GE 6SN7GTB "A"	GE 6SN7GTB "B"	140
Philips 5814A/12AU7 "A"	Philips 5814A/12AU7 "B"	129
Sovtek 12AX7WA "A"	Sovtek 12AX7WA "B"	547

mode they are very close to the specified value. The ADC in the display must cover a very large voltage range—400V down to millivolts—and the gp signal for pentodes is less than 50mV. I did not run the EL34s as triodes, due to the lack of published gm, rp, and µ specs.

COMPARISON WITH AUDIOMATICA SOFIA DATA

I also measured the dynamic characteristics of the same tubes using the Audiomatica Sofia[3] vacuum-tube curve tracer, in order to compare its measured data with that of the VacuTrace. This data is listed in *Table 3*.

The first issue is that the 6L6 is not in the Sofia tube list, and I didn't wish to use the 6550 program for fear of overpowering the 6L6s. I e-mailed Audiomatica to see whether they had a software update, since the KT66 is listed on their website. They said to use the EL34 test program for the 6L6s and leave out the suppressor connection.

Figures 14 and *15* are the Sofia curves for the two 6L6s in pentode mode, using the EL34 test program. They are displaced about the same as I observed with the VacuTrace. *Figure 16* is the triode mode curve for 6L6 sample #1. I also ran sample #2, and found the two 6L6 triode curves to be almost identical.

Figures 17 and *18* are the Sofia curves for the matched pair of EL34s (#1 and #2) in pentode mode. The Sofia also has an ultralinear test mode, but I did not use it since the VacuTrace doesn't have an ultralinear mode per se. I suppose you could adjust the screen voltage to the proper fraction of plate voltage in hold mode, but I don't know how you could do that dynamically for the sweeps.

According to the Sofia ERR matching software, these "matched" tubes had the second worst matchings in the test—go figure! The plate current at a given grid voltage is much higher in the #1 sample, and the curves diverge as the current increases.

Since I had two additional Mullard EL34s (not matched), I decided to run those on the Sofia as well. These two tubes have plate curves that are much closer than the "matched" pair. The data is shown in *Table 3*, although the curves themselves aren't.

On to the dual triodes. *Figure 19* shows the Sofia curve for 6SN7 section "A." The curve for section "B" is essentially identical in this G.E. NOS sample. Likewise with the 12AU7 (see *Fig. 20* for the Sofia curve for section "A"). There is a slight bit of divergence as the grid voltage goes below the −16V grid voltage curve.

Finally, I tested the 12AX7. Its two sections aren't well matched at all, so I show both curves (*Figs. 21* and *22*). This poor showing is reflected in the high ERR of 547 in *Table 4*.

I ran the Sofia matching program for all the tubes in the test (except the 5Y3), including the two additional random-sample Mullard EL34s. ERR is a dimensionless figure of merit representing the smallest root-sum-square differences between the measured values of plate current and plate voltage over the specified range of grid-bias values. The lower the number, the better the match. An ERR of less than 100 is a very good match. The data is shown in *Table 4*.

CONCLUSION

The performance of the VacuTrace as a curve tracer will be limited by the analog oscilloscope used for display, which I described with my particular scope, a Tek 7603. However, a comparison of the VacuTrace data with that of the Audiomatica Sofia (almost three times the cost) shows reasonably good correlation, especially for the triodes. Once set up, it is very easy to test multiple samples of the same tube type. In pentode mode I found that small variations in the operating point with both curve tracers could produce large parameter changes, especially in gm.

The VacuTrace is easy to use and a real value...and a kit version is available to boot!

Manufacturer's Response:

I'd like to thank Mr. Hansen for his review of my VacuTrace. Producing such a detailed and thorough article requires much effort.

It might help to elaborate on a few points made in the article. First was the author's discovery that his "lab matched" tubes were actually not. This may happen more often than readers might expect. Single point matching can be inadequate. Personally, I believe the truest way to match tubes is the "overlapping curves" method used in the VacuTrace (or by other curve tracing techniques).

Mr. Hansen noted that measured tube parameters for pentodes are very sensitive to changes in operating point. This is indeed the case, since pentodes have very high gain and are quite nonlinear. You won't see many pentode output stages operating without feedback! It was also seen that the VacuTrace readings for rp (1/gp) were a bit on the high side. This might be caused by the fact that VacuTrace does not separate out the screen current.

Finally, diode curves were found to have an oscillation causing the displayed curve to be very wide. I discovered this to be due to the P4/P6 switch, which selects only one anode at a time. Floating an anode does not make the tube happy. When both anodes were tied together, the displayed curve became very clean and thin. I will be removing this switch from future production.

❖

Jim Hagerman
President
Hagerman Technology LLC

REFERENCES

1. "Semiconductor Tester," Hansen, C., *Popular Electronics*, May 1999, pp.31–41.

2. None of the hobbyist "digital scopes" that I am familiar with and use PC soundcards or PC interface adapters have X-Y capability. Perhaps readers know of such a device. The clarity of the display on an analog scope decreases as the number of sweep curves increases. This might be improved by digital storage, if available in X-Y mode.

3. "The Sofia," Don Jenkins, *V&T News* 1/99, pp. 11–19.

Xpress Mail

VIDEO SPEAKERS

I have a pair of Polk Audio bookshelf speakers. I want to put them in a wall unit system along with my 27″ television. My problem is when I set the speakers next to the television, it affects the picture tube on the TV. Corners of the tube get a dark shading.

I am told that the speakers are not shielded. Can this be remedied somehow without purchasing new speakers? Can they be shielded reasonably easy and cheaply?

Greg Rauch
Mid-Continent Product Analyst—
Heavy Fuels
E-mail: grauch@Tosco.com

Edward T. Dell responds:

The quick answer to your question is "No."

There are two ways to adapt regular speakers for use with TV. Shield them with MU metal, which is nearly impossible since the material is expensive, difficult to work, and ineffective if not handled just right.

Speakers for such use are made for the purpose and include a bucking magnet, usually the same size as the speaker's magnet, thus killing the radiation. These are often also provided with steel shields as additional insurance.

Best answer, sell your Polks and purchase some made for video compatibility. For home theater, only the center-channel speaker and the two fronts need to be shielded. Use our "Yard Sale" column to look for replacements and sell the Polks.

DC/DC CONVERTERS

I recently saw your article by Jules Ryckebusch about how he utilized a DC/DC converter to deliver ±15V for the audio circuit from the auto's 12V source (Sept. '01 *aX*, p. 8). If it is of any help, I am a Canadian distributor of DC/DC converters for audio/video use. I have 1W to 15W, 33mA–500mA per rail, wide input from 9–18V. These units all supply ±15V out.

Should your readers in NA/Canada need them, they may contact me directly or go to our web page at: http://ssc_electronics.tripod.com/SSC_ELECTRONICS/ Hope this may be of help to your readers.

James Kay
SSC Electronics
Toronto, Canada

...AND MANY MORE

Congratulations on the completion of your first anniversary. The new magazine is more attractive to me than the separate, more specific, predecessors. A wide coverage of audio themes is probably more interesting in the long run. Thank you for your spirit of improvement, and for striving to perceive the pulse of enthusiasts.

Carlos E. Bauzá
Guayama, PR

DAT HELP

My father, Pat Oppermann, was a devoted member of SMWTMS (Southern Michigan Woofer & Tweeter Marching Society) and was as enthusiastic about his membership there as any other group he was associated with. Members were kind and helpful when it came time to find homes for some of his recordings and equipment. I am now asking for another favor:

His proudest possession at the time of his death four years ago was his Panasonic SV 255 DAT recorder. He used it many times to record my brother's and my band, the New Reformation Band. I would like to continue to use the recorder for the same task.

My problem is that I can find no power source—either battery or AC adapter. Nor do I have an instruction book. The good people at Panasonic have declared the machine obsolete. "After all, Mr. Oppermann, that recorder was made back in 1989!" Oboy.

Anyway, my dad knew his recording gear pretty well, and he sure thought a lot of his SV 255. I hate to discard it. I'm sure it's a fine piece of equipment, but I don't know where to go next. Perhaps you can offer me a tip or two.

Nick Oppermann
1893 Avalon
Saginaw, MI 48603
(989) 797-2193

DRIVER PARAMETERS

I read with great interest Joe D'Appolito's review of the Speaker City MTM-18 kit. I noted your belief that the internal volume of the speaker was too small. I am interested in using these same drivers in a design of yours, the 717 for A&S. I computed the volume of the 717 to be approximately 33 ltrs without bracing and other losses. This seems close to the MTM volume.

In addition, I have built an ARIA 5 with the Raven, which seems to exhibit the same problem at the bottom end as the MTM-18 and has less volume than you seem to recommend for the MTM. I know that you can't give individuals speaker-building advice, but since you seem to give contradictory volume recommendations, I wondered if you can clarify this.

Harold Goldman
LINHARSETH@aol.com

Joseph D'Appolito responds:

The A&S 717 was designed in the early '90s. Parameters for the Scan-Speak 180mm woofer have changed substantially since that time. In particular, V_{AS} *has increased greatly, requiring a larger enclosure.*

I am not aware of any low-end problem with the ARIA 5 as I designed it, but, again, driver *parameters have changed over the intervening years. I have not made any update to the enclosure design. It is quite possible that the volume originally specified is no longer valid.*

AC LINE CORRECTION

In my article on power line noise ("The AC Power Line and Audio Equipment, Part 2," Oct. '01 *aX*, p. 66), I made a wrong attribution of Rick Miller's sidebar to Walt Jung. Reference 4 on page 73 should read:
4. R. Miller, "Measured RFI Differences Between Rectifier Diodes in Simple Capacitor-Input Power Supplies" sidebar, *TAA* 1/94, pp. 26-27.

Charles Hansen
Ocean, NJ

OMISSIONS

You list new products at the front of your magazine but never give the prices. Other mags almost always provide that info. Also, your review of the Tannoy S8LR (Sept. '01, p. 58) did not include a photo, any dimensions, driver sizes, manufacturer's specs, or prices!

Leo Mack
LeoRobertMack@aol.com

Thanks for the heads-up on these issues. We have not included prices in our new products section because the Postal Service sometimes reclassifies such pages as ads and charges higher prices for delivering them. However, your criticism of the lack of information on the S8LR is well taken. This was an oversight on our part. Such information normally appears in our reviews—and will in the future. A photo of the Tannoy (Photo 1) *and its specs* (Table 1) *accompanies this letter.—Ed*

TABLE 1
SPECIFICATIONS FOR THE TANNOY S8LR

PERFORMANCE	
Recommended Amp Power Watts Per Channel:	10–20
Peak Watts Power Handling:	220
Sensitivity (2.8V@1m):	90dB
Nominal Impedance:	8Ω
Frequency Response ±3dB:	50Hz–20kHz
DRIVER CONFIGURATION	
Duralumin Tweeter Size:	1"–25mm
Polypropylene Mid-range/Bass Driver:	8"–200mm
CABINET	
Enclosure Type:	Front Ported Reflex
Weight:	27.5 lbs–12.5kg
Dimensions H × W × D:	17.7" × 9.5" × 11.5"

All models are finished in black ash vinyl with pewter grey front baffle or optional finish of cherry vinyl with champagne front baffle.

Tannoy/TGI North America Inc., 335 Gage Ave., Suite #1, Kitchener, Ontario Canada N2M 2C8, (519) 745-1158, FAX (519) 745-2364, website: www.tannoy.com.

WHAT'S NEW?

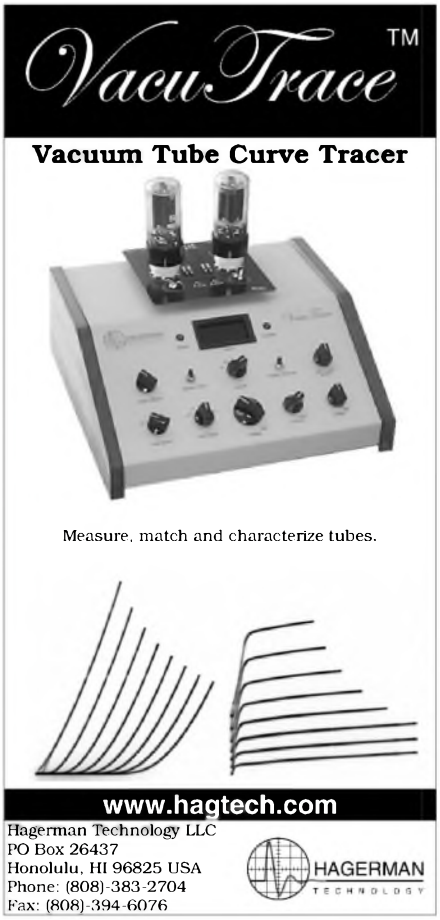
About the August 2001 issue:

1. Please note that the woofer used by the author in his electrostatic speaker ("Electrostatic Odyssey," p. 40), Dynaudio, has been withdrawn from the DIY market for close to one year.
2. I do not understand why you continue to review fully assembled units such as the Caspian and Sony CD players (pp. 58 and 66). I can read those kinds of reviews anywhere. Kit reviews would be much more appropriate for this magazine.
3. You should also indicate in construction articles when Old Colony will be offering the PCBs for a project.
4. You should rename your "New Chips on the Block" column, because you have reviewed chips that have been on the market for 5–10 years in the column. "Chips on the Block" would be much more accurate.

Claude Dickson
bogieworf@hotmail.com

Thanks to Mr. Dickson for his comments. Manufacturer discontinuance of drivers is one of the hazards of writing and publishing these days. I leave it to readers whether our reviews of non-kit items are equaled else-

PHOTO 1: Tannoy S8LR.



where. Any project where a circuit board is available is always mentioned in the article.—Ed.

Charles Hansen responds:

Thanks for your input. I believe there were two instances of existing chips we mentioned in the "New Chip" column. In one case, a National power amp IC, we mentioned that it was an older chip. The intent was to point to the data sheet, which had been updated.

Another instance was an audio op amp for which a quad version was issued. We included the single and dual in the article because they are all on one data sheet.

We are also playing catch-up to some devices in the last two years, since the column is fairly recent.

While we make every effort to list all the new audio devices, the advent of 24/96 and 24/192 ADCs, DACs, and Codec chips has produced more than we can possibly cover at 2–3 chips per month. There is going to be a delay in the mention of some devices as a result. We do try to publish them in the chronological order in which we receive the manufacturer's press releases.

SUBWOOFER RESPONSE

The September 2001 issue includes a subwoofer design by Jules Ryckebusch ("Build an Automotive Sub-Satellite Speaker System," p. 8). This is a simple closed box driven by two identical drivers in an "isobaric" configuration. The design caught my attention because of its relatively large front chamber, roughly half the volume of the rear chamber. Remembering earlier experiments with computer modeling, I suspected that the actual response of this arrangement might deviate appreciably from the author's Butterworth high-pass design goal.

Sure enough, my computer analysis predicts that frequency response (bold curve in *Fig. 1*) includes a 3dB shelf at about 150Hz. For use as a subwoofer, this may actually be beneficial. Also, the low-frequency cutoff is a little lower than expected, and this is definitely beneficial. However, these characteristics are accompanied by a 2dB difference between the two cone excursions below 150Hz. At very low frequencies, maximum output will be limited by the cone travel of the front speaker.

The frequency-response glitch has been noticed previously, but usually ascribed to an organ pipe resonance in the space between the two speaker cones. This is obviously not the case here because such a half-wave resonance would occur at about 2kHz. The reason for the unwanted interaction has to do with loose terminology. To be truly isobaric, the low-frequency response of the rear speaker and rear chamber would need to be the same as that of the front speaker on an infinite baffle. This might be accomplished through a choice of speaker parameters or some kind of active feedback scheme, of which several patented designs exist.

When two identical drivers are used, usually bolted face to face, it is assumed that the resulting air cavity is stiff enough to rigidly couple the cones together through the frequency range of interest. If the cones are loosely coupled, then response variations such as those described can be expected to occur.

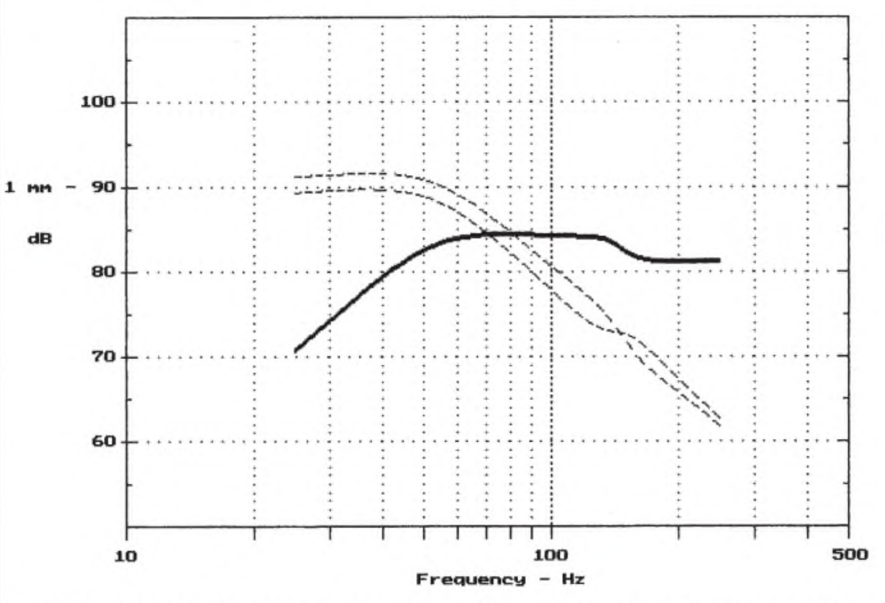


FIGURE 1: Non-vented isobaric woofer system—1W response (bold) and cone excursion.

G. L. Augspurger
Perception Inc.
Los Angeles, CA

Jules Ryckebusch responds:

Mr. Augspurger makes a couple of very valid points. Let me first say that I am a little surprised and honored that someone took the time to computer-model my subwoofer design. I knew that the "classic" design has the two drivers bolted face to face. I also know from experience that there is a lot of room for variance in box volume of up to 10-15% without drastically altering the design outcome of the enclosure.

I have seen other designs for isobaric enclosures that use a short piece of tubing the same diameter of the driver to separate the two drivers. I had considered this but could not readily find a short piece of what I needed. So, I opted to try not using this. My results audibly well met my expectations for using a pair of less than $20 apiece drivers.

What I will do is to try to fill in the void space between the drivers to get down to a short 8" cylindrical extension between the drivers. This will drop the front volume by almost 75% (assuming a 10 in^2 space reduced to an 8" circular space). I will try this and provide feedback on my findings. Once again, thanks for the observations and feedback. I plan on trying a 12" isobaric design very similar to this one for my home theater and will incorporate your input into the design.

PORT SIZE

Thank you for your article on port nonlinearity ("How Good Is Your Port?" Sept. '01, p. 24). This is a very important subject, which has received little attention. I'd like to add a few angles to what you said.

First, you concluded that even a 6″-diameter vent suffers significant compression at high power levels. The compression problem is greatly compounded by the fact that at the tuning frequency f_b, the output of the vent should be 3–10dB greater than that of the driver. In an "idealized" lossless system, the driver's displacement converges towards zero and the vent delivers 100% of the output.

In the real world, if the driver is moving a large distance at f_b, then the smaller diameter port is attempting to displace several times as much air. This means very high velocity and lots of compression in a typical vent. Therefore, the benefits of the ported box are greatly reduced. Tapering the edges of vents, and so on, is helpful, but by itself falls far short of solving the problem.

Note that the disadvantage of unloading below f_b is still not reduced much by port compression. Compare this to an acoustic suspension speaker, which does not have a port compression problem because it doesn't have a port, and also is loaded at low frequencies. Port compression seriously erodes the advantages of a vented design.

I know from modeling, building, and measuring transmission line speakers that there is no free lunch on the cross section of the line. If line cross section is less than the driver area, the sensitivity of the driver is reduced in the bass range. A 3″-diameter transmission line will produce serious problems for an 8″ woofer, regardless of any other dimension. This principle applies equally to vented speakers.

It naturally follows that large diameter vents are necessary, which are impractically long. This leads me to conclude that for SPLs much above 90dB, reasonably sized ports are simply inadequate. Passive radiators are often the only practical solution.

Almost all vented speakers suffer from this problem—to the extent that most people have never heard a vented system that does not have significant compression at high output levels. This leads to the widespread belief that vented speakers have mushy, sloppy bass. That's because they do.

Vented speakers that fully address this problem are very impressive at high power levels. The bass is solid and powerful, not sloppy. Those who take the appropriate measures to eliminate port compression (using passive radiators or very large diameter ports) will be amply rewarded for their efforts!

Perry S. Marshall
ebiz@futurezone.com

Bohdan Raczynski responds:

Thank you for taking the time to read my article on port nonlinearity. Indeed, the performance of loudspeaker systems at higher power levels in general, and port compression inclusive, have only recently been given some attention. Looking at the size of the ports in some commercial offerings, you would be forgiven for concluding that many designers have been blissfully unaware of the port compression problem.

Large signal analysis of loudspeaker systems often leads to the conclusion that two vented systems looking equally good at 1W power level will deliver rather different performance at 200W of input power. There are three major factors contributing to this phe-

nomenon: magnetic motor nonlinearity, cone suspension nonlinearity, and port compression. Fortunately, some knowledge on these issues has been disseminated in recent years and appeared in readily available literature. I would be hopeful that a growing number of professionals and hobbyists are now recognizing the need for moving beyond the "linear models" of the loudspeaker systems.

I would also like to thank you for your comments, based on your practical experience with large diameter ports. My observations would be the same. It is, however, a challenge to execute the large diameter port design properly.

MOSFET SOURCE

In response to a reader's "Help Wanted" letter, the speaker systems called "tombstones" are in *SB* 1/94, "A Full Range Open Baffle System," p. 30, by Warren Hunt and Joseph Janni.

Thank you for printing my letter about Power MOSFET availability problems ("Throw-Away Culture," p. 78, Sept. '01 *aX*). Mr. David W. Platt of Springdale, Pa., was kind enough to send me the phone number of the Farnell Co. (800-718-1997), and their part numbers for the SJ162 (355-926) and the SK1058 (356-001). Farnell is evidently the commercial only wing of Newark and they couldn't do business with me, but they gave me Newark's number (800-463-9275). Newark could go from the Hitachi numbers to their numbers, SJ162 = 08WX6470 at $12.05 each, and SK1058 = 08WX6482 at $11.06 each. These are special order items—no cancels, no returns, four weeks delivery, though mine came in less than two weeks.

So, thank you for your magazine and your assistance.

Art Day
Kingston, Wash.

DIY DAC

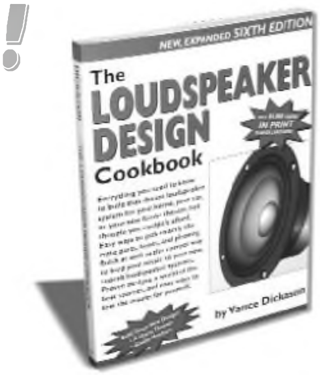
I am writing in regard to the excellent article on comparative DAC audio chips in the May '01 issue of *audioXpress* ("Exploring the DAC Chip World," p. 52). I have purchased the major parts for the Crystal 4390 project and am now laying out the work, but I have a few questions. I am just beginning my DIY studies, so please bear with me if these questions seem simplistic.

1. The data sheet for the LL1527XL transformer from the Lundahl website shows four possible ways to connect the output transformer (serial-serial, parallel-parallel, parallel-series, parallel-split). Which is the proper connection for this circuit?
2. In Fig. 2 there is one ground shown for the power supply. Does this ground simply terminate at the negative pole of the evaluation board or is it taken to another location? What is the size of the fuse F1?
3. In Fig. 1 there are grounds shown for each channel on the primary and secondary sides of the LL1527XL transformer. Do these simply connect together at the negative side of the RCA connector or are they taken to other locations?
4. What is the voltage rating and manufacturer of the 22μF capacitor that bypasses the 10k resistor on the primary side of the LL1527XL transformer?
5. Finally, is the metal case completely isolated from all power supply and audio circuitry boards or grounded to the power cord ground or perhaps another location?

Mark Vaughan
Vancouver, B.C.

Andrea Ciuffoli responds:

1. Both primaries and secondaries are in serial connection, and the central of the primaries is connected to the 10k resistor.
2. The ground of the power supply terminates at the negative pole of the evaluation board.
3. Yes, they must simply be connected to the negative with all the ground.
4. The best component that you could use is the 16V ELNA Cerafine.
5. You should connect the metal case to the power supply ground. Connecting the power cord ground to the audio circuit board could give noise problems if the other components of the system (CD, amp, preamp) are also grounded. In fact, some commercial hi-end products such as CD and DAC have a switch to isolate the power supply ground. It would be simple for you to add a switch for this purpose.



DVD ADVICE

Regarding Terry Ruprecht's "Help Wanted" letter (Sept. '01 *aX*, p. 84), in which he has no bass sound from his sub using a DVD-A player: He has run headlong into the "bass management" deficiency between these players and A/V receivers. DVD-A discs do not have a discrete bass channel. The current generation of DVD-A players does not include bass management capability that is compatible with most multichannel A/V receivers.

Even though his sub channel indicator lights, this does not mean that there is any audio in the sub channel—only that space has been allocated in the digital data for the sub signal. If the recording engineer didn't put an actual LFE signal on the DVD, there will be no bass from the sub.

There have been a number of workarounds suggested on the internet: setting the LF and RF A/V processor to "large" and feeding the front channels to the sub's speaker level inputs (if available), and so on. Outboard bass management units are also in the works, to add to the "kludge" factor. If you haven't already plunged into DVD-A, it might be advisable to wait until the DVD player and multi-channel A/V receiver folks have properly sorted out this problem.

Charles Hansen
Ocean, NJ

SIMILAR CIRCUITS

I saw the schematic for the Pilot 264 in the Aug. '01 issue (p. 87). My Fisher 500C receiver has a somewhat similar circuit, and both companies were in Long Island City, NY.

By grounding the 4Ω taps and connecting a center-channel speaker between the common of one channel and the 16Ω tap of the other channel, the speaker will get the sum of both channels. Fisher said this was for three-channel stereo or remote monaural, and recommended a 16Ω speaker. I think the 15Ω resistor and .1μF capacitor on the output of the Pilot was used to roll off the high-frequency harmonics.

Don Passantino
Elmhurst, N.Y.

HELP WANTED

I am looking for information on how to build my own speaker distribution panel. A panel that I am interested in is at the following site: http://www.smarthome.com/8267.html. I would like to know what the resistors would be and what I could use for the switches.

John Saxon
saxonjohn@home.com
Calgary, Alb.

Can you help me find information on a loudspeaker driver called the Dyan-pleat? Is it still being made? Where can I locate a source?

By the way, I enjoy your magazine!

Barry Stephen Goldfarb
Imbsg@prodigy.net

This is obviously a trade name, and we have a dim memory of it here on staff. Perhaps a reader can help.—Ed.

I am not a hi-fi buff, but for many years have possessed a very adequate system, topped with a very old, but nevertheless very good, pair of Mission 700 speakers. However, over the years the foam on the front of these speakers has disintegrated, and so far I have been unable to find anyone who does replacement foam. Please can you help? ❖

Geoff Curnock
geoff@the-reggies.freeserve.co.uk

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

Classifieds

VENDORS

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Yard Sale

FOR SALE

Rega Planet CD Player, as new, $500; North Creek Borealis 2 in custom floor-standing cabinets, also as new, $700. Ken Martinez, 505-280-8829 or e-mail kennethmartinez@earthlink.net.

JBL model 2345 1″ horns, $65/pair; Altec 806a drivers (need new diaphragms), $65/pair. Jack, 732-463-1472.

Nestorovic type 5AS full-range speakers: 4″ planar tweeter, 4″ hybrid dome midrange, new 8″ and 10″ woofers—oak finish, excellent condition, $1,500. Call 603-889-0728 for info and photos.

WANTED

Hobbyist seeking pre-recorded 2-track reel-to-reel tapes—7.5 or 15 ips. Can be 1950s/1960s commercial product or old studio dubs. Andy Pennella, 203-329-7498 or ajpennella@att.net.

Dynaudio 15W75 speakers. I need a pair in any repairable condition—will pay full price if perfect. Ernie Holmes, 916-726-7039, ernieh@lanset.com.

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CONTRIBUTORS WANTED!

AUDIO AIDS— Whether your thing is solid or hollow state. Crossovers, power supplies, or transformers. You have experience and know-how others could use. Why not share that information with *audioXpress* readers? We encourage short (several hundred words) tips and hints for other electronic DIYers and will pay a modest stipend for your efforts. Contribute to the wealth of information in this industry by sending your "Audio Aid," and any accompanying photographs or artwork, to:

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TEST TRACKS— What are your favorite music selections you use to test audio gear or to show off some special capability of your system? Share your favorites with others. Simply describe your seven top pieces (not to exceed 1000 words); include the names of the music, composer, manufacturer, and manufacturer's number; and send to the address above. We will pay a modest stipend to readers whose submissions are chosen for publication.

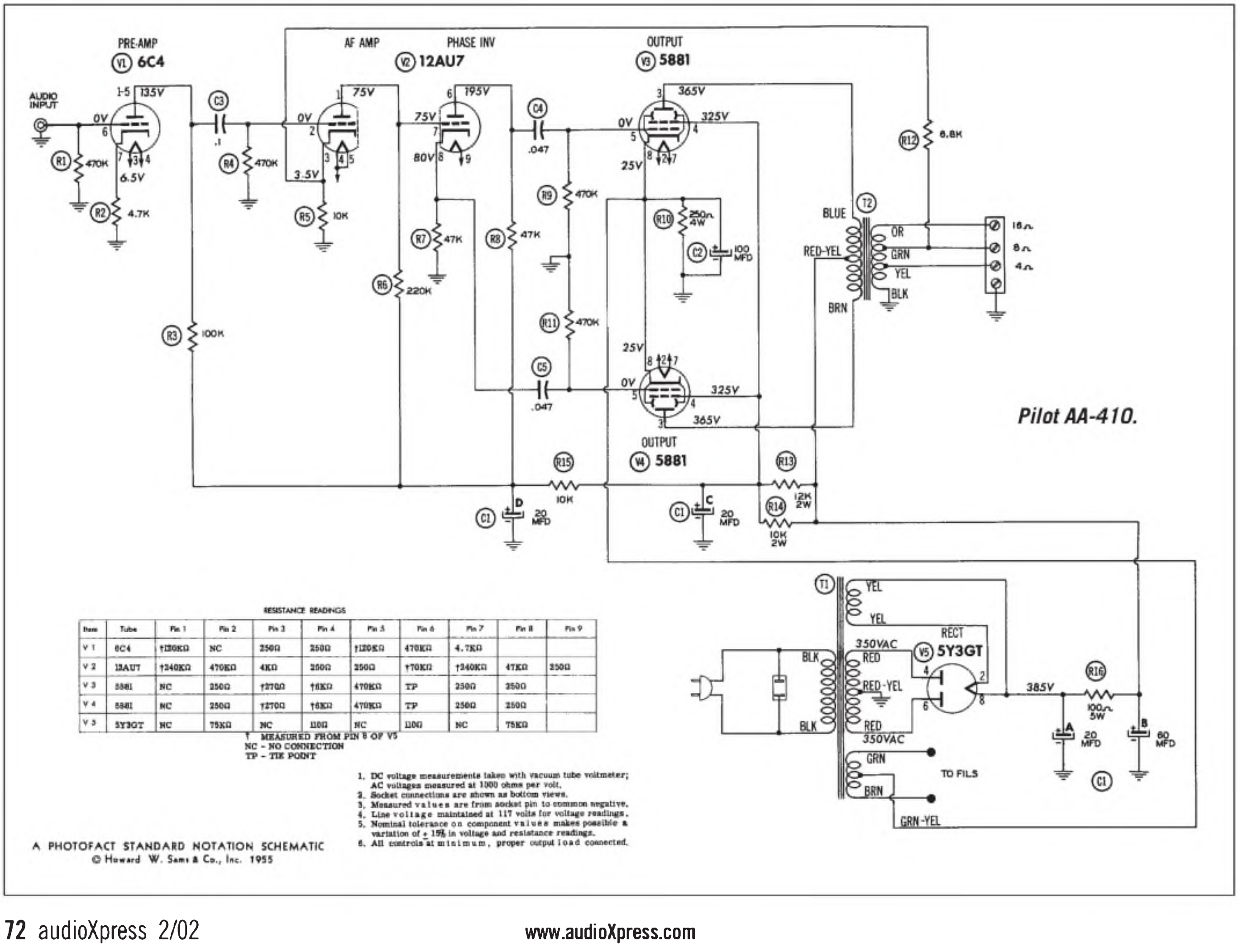
Classic Circuitry

Pilot AA-410

This simple push-pull circuitry features common tubes, easy-to-obtain transformers, no adjustments to mess up, plenty of power, and an attractive, simple chassis layout. This amp is perfect for a starter.

Beginners take note, don't let the 6C4 tube shown in the schematic stop you from building this amplifier. The 6C4 is electrically identical to a 12AU7, which has two identical sections, the 6C4 only one. Using the 12AU7 pinout shown, this amp can be built using half of a 12AU7 instead of a 6C4. In fact, a stereo version could be built using a single 12AU7 replacing the 6C4s for both channels. ❖

Lance Cochrane
San Francisco, Calif.



Pilot AA-410.

RESISTANCE READINGS

Item	Tube	Pin 1	Pin 2	Pin 3	Pin 4	Pin 5	Pin 6	Pin 7	Pin 8	Pin 9
V 1	6C4	†120KΩ	NC	250Ω	250Ω	†120KΩ	470KΩ	4.7KΩ		
V 2	12AU7	†240KΩ	470KΩ	4KΩ	250Ω	250Ω	†70KΩ	†240KΩ	47KΩ	250Ω
V 3	5881	NC	250Ω	†270Ω	†6KΩ	470KΩ	TP	250Ω	250Ω	
V 4	5881	NC	250Ω	†270Ω	†6KΩ	470KΩ	TP	250Ω	250Ω	
V 5	5Y3GT	NC	75KΩ	NC	110Ω	NC	110Ω	NC	75KΩ	

† MEASURED FROM PIN 8 OF V5
NC - NO CONNECTION
TP - TIE POINT

1. DC voltage measurements taken with vacuum tube voltmeter; AC voltages measured at 1000 ohms per volt.
2. Socket connections are shown as bottom views.
3. Measured values are from socket pin to common negative.
4. Line voltage maintained at 117 volts for voltage readings.
5. Nominal tolerance on component values makes possible a variation of ± 15% in voltage and resistance readings.
6. All controls at minimum, proper output load connected.

A PHOTOFACT STANDARD NOTATION SCHEMATIC
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