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Behind the Scene

Dr. Joseph D'Appolito has been working as consultant for Usher Audio since early 2000. A world renown authority in audio and acoustics, Dr. D'Appolito holds BEE, SMEE, EE and Ph.D degrees from RPI, MIT and the University of Massachusetts, and has published over 30 journal and conference papers. His most popular and influential brain child, however, has to be the MTM loudspeaker geometry, commonly known as the "D'Appolito Configuration," which is now used by dozens of manufacturers throughout Europe and North America.

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

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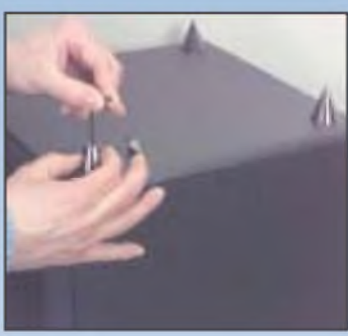
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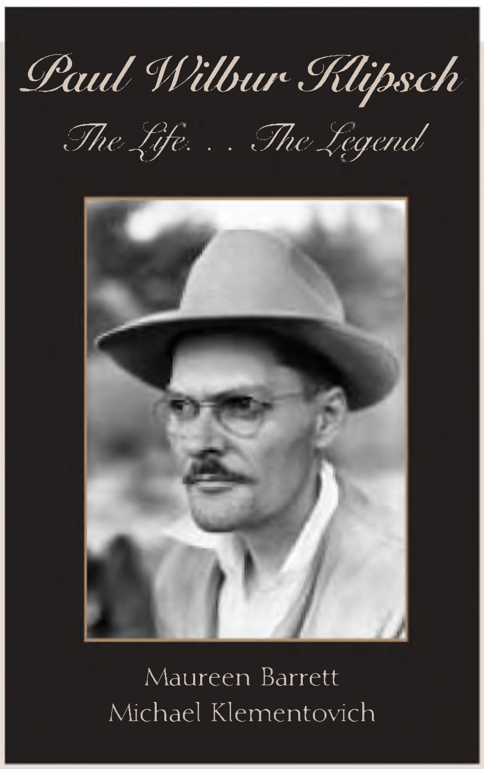
What do Thomas Edison, Dr. Jonas Salk, The Wright Brothers and Paul Wilbur Klipsch have in common?

They all share space in the Engineering and Science Hall of Fame. Space set aside for only those who in one way or another changed our lives and improved the quality of life for all humanity.

Paul Wilbur Klipsch



With little more than an advanced degree in electrical engineering, unorthodox ideas about acoustics, and a workshop so tiny that two people could barely squeeze inside, Paul W. Klipsch founded a company to make loudspeakers in 1946. Today, Klipschorns are among the most prized state-of-the-art speaker systems, primarily due to the vision and rigorous scientific standards of Klipsch himself. In *Paul Wilbur Klipsch: The Life. . . The Legend,* authors Maureen Barrett and Michael Klementovich have created an in-depth exploration of this American visionary in the only authorized biography of Klipsch to date.

Klipsch has been a ballistics expert, serving his country in World War II, a geophysicist, an amateur pilot, and a determined suitor. In this biography, we get not only a window to his scientific genius, through original articles, transcribed lectures with original drawings, notes, and the text of his twenty-three patents, but a glimpse of his original personality, especially encapsulated in the special bond he has with his wife, Valerie. Their lively relationship is the foundation for many of the anecdotes that Barrett and Klementovich use to illuminate Klipsch's distinctive way of thinking about the world around him.

Paul Klipsch's achievements have been honored with awards too numerous to name; he has multiple honorary degrees and has been enshrined into the Engineering and Science Hall of Fame for his contributions to the fields of acoustics, balltics, and geophysics. Additionally, he has been a generous benefactor, contributing his time, energy, and money to better educate America's youth, and to further scientific understanding. Most of all, he has created an enduring legacy in the minds of grateful audiophiles. *Paul Wilbur Klipsch: The Life. . . The Legend* captures the essence of a unique intellect: a pioneer, an inventor, a husband—a modern Renaissance man.

Paul Wilbur Klipsch The Life. . . The Legend can be ordered by calling the publisher, Rutledge Books, Inc., at 1-800-278-8533.
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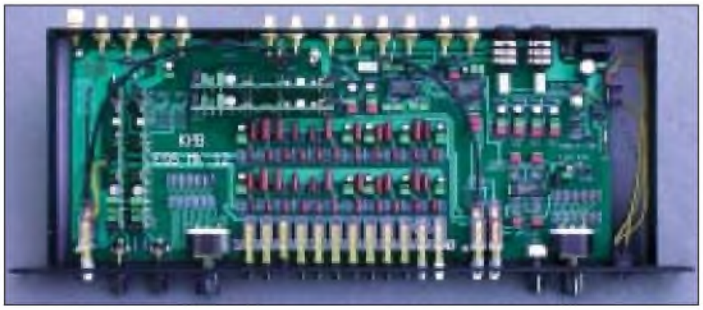

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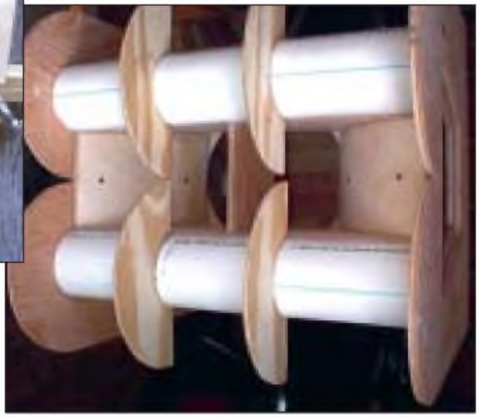

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The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it.

JOHN STUART MILL

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Bass and the Room

Balance of treble and bass in your speaker system may have more to do with what materials were used in building your house than other factors—which is good news for audio amateurs. **By Martin Colloms**

Many articles and lengthy discussions dominate the speaker design arena on the subject of low-frequency performance. While I would not want to sideline the value of absolute maximum loudness as a primary criterion of a high-quality speaker system, it is in the bass where the perceived horsepower of a speaker resides. Powerful, clean bass has the power to surprise, even shock, the listener. Bass underpins the sound reproduction and, to a great degree, contributes to the sensation of power. Clean, extended bass provides a sense of scale and reinforces the quality of tonal balance heard through the midrange.

There's endless debate concerning the merits of various bass alignments; I've tried many and assessed innumerable speakers with many more for publication. One lesson I learned early on concerned the innate conservatism of good engineering. A well-made design should embody a vital and effective balance of three parameters, namely, performance, reliability, and economy. Unless price is no object, every designer must fix on a realistic target for an affordable bass driver, and this choice needs to be consonant with the allowable size of the planned enclosure needed to house and properly load it acoustically.

ABOUT THE AUTHOR

Based in London, UK, Martin Colloms became an independent electroacoustic engineer and technology advisor in 1975, founding his own evaluation and review laboratory after a relatively brief affair with the design of high band radio telecomms and GHz sampling oscilloscopes while graduating and working as a design engineer. He had thought that Hi-Fi and review writing was not a real job, but it eventually took him over. A particular love of music reproduction and loudspeakers soon led to *High Performance Loudspeakers*, which is still alive and well in its 5th edition under the publishing care of John Wiley.

The two themes of music in all its forms and the faithful reproduction of sound remain paramount for Martin. With literally thousands of published technical reviews down the line, he comments that the exploitation of a solid theoretical and evaluative platform has still to provide unambiguous explanations for many audible differences in sound quality observed for well-specified audio equipment. The search for an understanding of these issues remains a source of fascination, a field open to amateur experimenters and professionals alike.

COMPUTER-AIDED LOW-FREQUENCY SYNTHESIS

Long before this decision is made, the entire low-frequency scenario deserves consideration. Much teaching and literature indicates that of the skills which may be applied to synthesize a well-balanced, timbrally neutral loudspeaker system—one which is truly satisfying to the ear, and sounds natural at a range of sound levels—it is those for the bass region which promise precise design. Thus we are generally taught to employ the popular lumped parameter enclosure model, apply the Thiele/Small driver specifications to deliver the chosen response, and we are also asked to assume that our listening environment and enclosure behavior are together valid for a 2 pi space (a radiation hemisphere equivalent to a plane infinite baffle) over the whole low-frequency range.

Relatively simple computer programs readily allow for rapid synthesis and usefully reveal many theoretical possibilities of enclosure and driver combination. [Significantly, the more advanced programs allow local environment boundaries to be dialed in (e.g., AKABAK by Joerg Panzer), and these deserve greater attention.]

We really do need to be cautious at this point. That small print warning that "these synthesis calculations assume 2 pi space" really needs thinking through. While convenient for the design of the computational model—this relying upon a diffraction-free, reflection-free environment—real life is much more complicated. The many complexities of the substantially uneven reverberation of listening rooms at low frequencies indicate that a more worldly approach is required. Issues such as path or integration delay between the port and main output for larger speaker enclosures will not even begin to figure in this commentary.

USE OF A TARGET FUNCTION

It's thus helpful to involve the concept of "target function," a term I first saw some 25 years ago in a paper by KEF engineers concerning crossover design. The design method outlined looked at the acoustic response of the driver, crossover, and enclosure as a complete coupled system, and considered the required "target function" for design, aimed at the deliverable acoustic output. It moved forward from the previous almost slavish subscription of designers to textbook crossover alignments involving idealized drives.

Applying this concept for the bass range, we should include the effect of typical speaker location, and the resulting low-frequency room loading, as parameters for defining the target function. Many believe that too great an emphasis has been placed on 2 pi design for the low frequencies. Convenient and comforting as such software based design resources may be, in fact, they're really just a starting point.

EFFECTIVE BASS Q

An isolated 2 pi computed result for an intended alignment is simply unworldly. The actual bass alignment heard by the listener involves much more than this. To gain some insight, consider a speaker in which the bass alignment is actually matched satisfactorily to the radiation space. System Q_{TC} is rightly seen as a powerful parameter in the set of variables, and if too small, is responsible for bass "dryness" on the one hand, and if too large, "boom" on the other. Depending on the class of alignment, a particular Q_{TC} value is generally argued as being just right, though in truth personal taste does come into it.

Intriguingly, and regardless of room considerations, I will now show how it's possible to change the effective, perceived bass Q without changing any of the low-frequency parameters at all. For a given two-way system, to increase the effective perceived bass system Q, all I have to do is put a resistor in the feed line to the treble driver, and at my discretion, mildly increase the lead inductor of the crossover to the bass driver to match the new reduced treble level and thus bring the system into balance.

The bass "weight" of this system greatly depends on the *in situ* frequency balance heard by the listener. Drop the mid and treble as suggested, and against a notional perception of a mid point "hinge" for tonal balance, typically 800Hz, the bass now sounds "boomier," weightier in perceived terms, in fact, with an effectively higher Q value.

Since the bass is the range where driver and box fundamentals dominate the behavior, and a redesign may be costly, it is a given among many speaker designers to sometimes respond to the criticism "there's not enough bass" by recommending moderate attenuation in the treble! How the bass actually "sounds" in a given arrangement involves a multitude of influences.

The adoption of a low-frequency target function encompasses many of these factors, and this is where the amateur may well have an advantage over the professional designer. The latter must somehow use his/her best judgment to produce systems in which the bass alignments are a fair compromise for a range of these local parameters—the listener's complex room acoustic, the speaker placement in it, and the preferred listener position. All these also affect the actual target Q heard at the listening position. By contrast the amateur may assess and learn his or her local acoustic and thus fine-tune a custom design for a more optimal result *in situ*.

EFFECT OF BUILD AND ROOM STYLE ON EFFECTIVE Q

Bass Q_{TC} targets vary among countries. A designer in Europe is generally working in Ferro-Concrete built apartments—"East Coast," if you like—and acoustically comparable stone and brick built multi-room houses. Conversely, stateside, out of town, the larger homes are frequently timber frame, drywall, and part masonry infill construction. In addition, the areas and overall volumes of such houses are substantially greater than in Europe, probably as a consequence of much lower land cost. Large open plan constructions in the US mean that the low-frequency acoustic loading the loudspeaker is very different from that in Europe.

Large timber-framed houses have structures where significant low-frequency energy leaks out and where the effective room volume is large, leading to reduced low-frequency gain. Neville Thiele confirmed to me that his seminal work on low-frequency alignments and design was based on 2 pi partly because the studio building where he researched and listened was a relatively "leaky" timber structure. "2 pi seemed about right," he said. Thus our standard calculation environment was born!

More enclosed, well-proportioned rooms are common in more traditional solid masonry buildings, and the bass response of the speaker there will be dominated by a room whose reverberant losses reduce substantially with frequency below 500Hz, leading to a spatially resonant behavior and real frequency averaged gains in perceived low-frequency power. Ideally, the speaker alignment will be over-damped relative to a 2 pi maximally flat alignment in order to try and account for the progressive room gain at lower frequencies.

Consider a design for a $2ft^3$ infinite baffle speaker with maximally flat, Q_{TC} 0.7 alignment and a –3dB point at a desirably low 30Hz, perhaps achieved at a considerable driver cost and probably with an overall efficiency compromise. It may be just right for an out-of-town open plan mansion, but will likely sound as though it has a boomy, fatiguing Q_{TC} of 1.5 when operated in the music room of a London town house. Data obtained from spatial averages on several such rooms for some 100 different speaker systems indicates a typical gain from this type of room of about 5dB by 30Hz relative to free-field reference band, for a near-to-wall placed system.

This (for this example an *in situ*) progressive low-frequency gain is roughly comparable to shift from a Q_{TC} of 0.7 to one of 1.5 and is of course equivalent to big changes in the bass magnet, for example; in this case about 40% of the effective flux. Factoring in the room match clearly can have a dramatic effect on the design target, the proposed alignment.

MAKING A START; WHAT ARE THE COMPROMISES?

A) Assuming we have a reflex design, what about the port?

Where there is no budget limit in design, we can choose large powerful drivers and expect good power handling, fine linearity, and a generous peak excursion capability. Such quality and performance ensures that over a sensible operating range of loudness and power, the bass quality—and the accuracy and stability of the intended low-frequency alignment—will not be significantly compromised. In this case what you design is more or less what you get.

For lesser mortals, smaller bass drivers and compact enclosures, with finite excursion and limited volume velocity from the port, are the norm. As the British put it for such a situation, "this is trying to get a quart out of a pint pot."

By way of example, take a 6.5″, or 170mm, driver in a compact 15 ltr or half cubic foot enclosure, ported. Cognizant of the port self-resonance aspect, which ideally indicates a maximum of 2:1 for length to diameter, a box tuning to a spectrally and musically useful 40Hz lower limit will theoretically provide an effective subjective performance down to 35Hz. One calculated port solution indicates a port diameter of 2.5cm for about 4cm of length.

This is nonsensical, because a 2.5cm port driven firmly to resonance at 40Hz would become turbulent quite early, then go into nonlinear overload and thus choke off the intended low-frequency box contribution before it is loud enough to be worth having. Here some compromise on the ideal port ratio is essential to meet the need, and a typical compromise is 4.5cm diameter, usable to around 90dB SPL, for which an 18cm duct length is required. Given its potential for a half wave pipe mode at about 970Hz from local midrange energy feed, the duct should be located to the back, with the internal entry displaced away from the main source, typically located behind the closed back treble driver.

While distortion and dynamic compression will be increasingly evident above 96dB, the generally lower level of low bass energy for most recorded programs means that the system as a whole will then manage to play to realistic and useful peak sound levels of 103dB and more.

B) Use of realistically derived T/S parameters

A key question concerns what alignment is aimed for in such a system, and at what sound level. We also must consider the sensitivity of the ear to bass level and distortion. There are absolute minimum sound levels, below which bass frequencies are just not perceived, so it's then questionable whether we should try to reproduce them. This goes to the very heart of engineering philosophy, and the result is not just a question of technical elegance, rather it hinges on effectiveness.

For smaller systems that need to work hard for their living, it is vital that the target function includes a design which is based on real worked T/S parameters, preferably measured with constant voltage, as they will actually be used, and at a power level where a representative part of the linearity curve for dynamic compliance is reached. Computerized measurement at unrealistically low drive levels—e.g., from computer card devices, and not when buffered by a more powerful test amplifier—can result in grossly unrealistic measured parameters leading to erroneous design.

C) Effect of crossover network

In very compact, low-efficiency systems a designer may achieve accurate tonal balance with three, even 4mH of associated series inductance to the bass-mid driver. Series resistance might account for 0.8 to 1.2Ω, depending on inductor design. These L and R parameters will significantly affect an intended low-frequency alignment and also need to be accounted for in the synthesis. While compensation for the motional impedance can help restore the original predicted alignment, the additional network required is large, adds expense, and in some cases may reduce sound quality. Many designers choose not to complicate the issue and go back to the target function approach for effective bass alignment, making adjustment at this point in view of the crossover influences.

D) Internal absorption and ideal stuffing

Many amateur and professional system designers aim for a given familiar alignment and then resort afterwards to some fine-tuning once the upper bass to midrange tonal balance begins to right itself during the design process. For a sealed box, choice of the type and amount of stuffing may help achieve this. Too little stuffing is generally considered better than too much, which in addition to lowering Q_{TC}, may also reduce the bounce and life in the bass; while it's worth noting that the intended energy absorption and damping doesn't just affect the fundamental bass range. It may also over-damp the lower mid, making the system sound lean and too dry. An over-absorptive acoustic loading inside a box actually reduces the mean sound level radiated at the front by up to 1dB, and in my book, every dB counts.

A guideline used by many designers subscribing to the "just right" school places the weight of stuffing at about 0.9 to 1.3 times the weight of the air enclosed. So for a 20 ltr 0.7ft^3 box, air at 1.2kg per cubic meter, you would use about 12g of stuffing per 10 ltr, or 35g per cubic foot. Much beyond this and the resulting system may sound rather staid.

The stuffing should always be mechanically stabilized so it cannot move about. If it does, the result is time-related nonlinearity, which may be audible as a poorer transient response at low frequencies.

E) Trial and error reflex box tuning

For reflex designs, within reasonable limits, an ad hoc variation of the box alignment is possible. This may be readily achieved and the results rapidly assessed by fitting a "trombone" style adjustable port, initially located on the outside of the box for convenient adjustment. Once determined, the port may be reversed, fitted to the inside, and the results verified for this condition. Some minor final adjustment may also be required.

For example, a given small system was built to provide a maximally flat 2 pi textbook alignment with the port tuning at 50Hz. Under room-loaded conditions, it suffered from some moderate bass excess; i.e., its target function has a higher than designed, effective Q. Carefully iterating the speaker location with respect to the primary local boundaries of the listening room—back wall, floor, and side wall—and with port length, while maintaining a sensible average loudness around 90dBA at the listening position using repetitive music and/or pink noise, the best compromise of overall tonal balance and low-frequency extension was obtained when this port was de-tuned to 43Hz.

While that 7Hz drop doesn't sound like much, this is not a trivial adjustment, and the 50Hz port of given diameter, 7cm long, now needs to be 12cm long. Once revised, this speaker sounded as though it had greater bass "punch," more extension (now down to 38Hz, room loaded) and was also more even, when assessed in 1/3 octave bands, spatially averaged.

F) Alignment variation with power

Now for this hypothetical 20 ltr box: Suppose that the bass range is now precisely tuned to a room, also compensated, matched, and aligned for the linear working power region of the port and driver. So far all is fine, and at moderate listening levels up to a few watts it sounds just right, which is actually fairly loud in practice. (You would not want to remain in the same room for long with 1W into a typical speaker fed 1kHz continuous tone.) Now turn up the volume, let's say to approximately double the perceived loudness, in other words, 10dB.

It's highly likely the system will not sound the same. There are several reasons for this. Your aural curve for sensitivity versus frequency is now on a different plateau for the higher loudness, and you may well hear more and deeper bass, if the speaker is up to it. However, that natural and satisfying growth of bandwidth and "power" with increasing level may not happen if the speaker begins to limit, as surely it will, sooner or later.

Given bass-rich programs, peak sound levels will first cause the port to reduce in effective diameter, de-tuning the box alignment as well as increasing damping due to losses. Eventually the port will become ineffective, only producing wind noise and higher frequency harmonics, moving the enclosure design almost to a leaky infinite baffle condition.

These factors will begin to cut substantial radiated acoustic power from the system in the 35–50Hz range. Now consider the driver, whose maximum extension, previously limited in this power band by intended resonant back pressure from the box interior, now begins to drive out beyond its linear mechanical and electromagnetic limits. Its Qe begins to rise sharply, and as the number of flux linked turns is reduced, it may begin to sound boomy at its upper, box resonance. Second and third harmonic distortion climbs rapidly; 10% and more is not uncommon.

Thus in addition to the loss of fundamental bass the speaker also shifts some of the energy up into the low midrange where aural sensitivity is greater, lending a "nasal" wooden quality to the sound and further reinforcing the impression of bass loss. Increasing overdrive will leave the speaker sounding "thin," possibly colored, with a forward, bright mid and treble. No wonder our natural reaction is to turn down the volume.

This behavior helps explain why many practitioners shy away from the bass reflex, which may become dynamically nonlinear when overdriven. Instead they seek safety in the relative simplicity of the sealed box, where driver and acoustic load parameters may be more stable, particularly when approaching overload.

THE REFLEX ADVANTAGE

So we must ask the question, "Why are so many of the most successful mid-priced speakers, as well as some high-end designs, given the vented treatment?" The answer is simple—it is that old virtue of engineering balance. For a given primary expenditure on magnet and box—if well designed and if the operating bandwidth is sensibly matched to the size and to the target loudness and driver/port capability—then a good reflex design will always play louder and deeper than the sealed box equivalent.

And there's generally another bonus with the reflex. Inspection of the driver excursion trends show that the mean displacement in the main bass power band for a good vented box design is substantially less than for an IB, sealed box equivalent, when radiating the same power and frequency range.

Less mean excursion means less average distortion, less current flowing in the coil/magnetic circuit, as well as better thermal behavior. You get lower compression and thus a more transparent and dynamic midrange if, for example, it is a two-way system with the usual bass-mid driver.

IB fans may castigate the reflex, usually citing a boomy-looking transient

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A High Performance Microphone Preamp

This article describes the design and construction of a very low noise microphone preamp. SPICE software was used to optimize the low noise design. By Ron Tipton

Recently I completed a project in which I measured the acoustical properties of several rooms using a personal computer with a high-quality sound card and appropriate software[1]. I was somewhat frustrated because my microphone preamplifier didn't have a low enough **noise floor** (see Glossary for unfamiliar terms) to give me as much **dynamic range** as I would have liked. So I developed a preamp with very low output noise, negligible **harmonic distortion**, and negligible **intermodulation distortion**.

INPUT STAGE DESIGN

The first step was a data book search to find solid-state amplifiers with a low input noise voltage (*Table 1*). The l/f noise, so-called because its spectral density increases as the frequency decreases, is a curve for each amplifier (*Fig. 1*). However, the "spot" value at 1kHz is a good indicator of a low noise device, as is the so-called "corner frequency," that is, the frequency at which the noise voltage sharply increases. A low corner frequency is better because there is less total noise within the audible range.

From *Table 1*, the SSM2017 looks like a good choice for the input stage, but unfortunately the manufacturer has discontinued it. The second table entry, the OPA227, looks pretty good, but I decided to first try to duplicate the input configuration of the SSM2017 with discrete transistors to see how well I could do. A data book search for high gain, low noise transistors led me to the 2N5087 (PNP), and 2N5088 (NPN), which I could use in a compound pair for much better linearity than I could get from a single transistor (*Fig. 2*). The 2N5087 data sheet[2] has a graph of wideband noise figure contours as a function of source resistance and collector current. A collector current of about 400µA looks as though it would minimize the noise in the source resistance range–150 to 600Ω–that matches most dynamic and buffered capacitor microphones.

ABOUT THE AUTHOR

Ron Tipton has degrees in electrical engineering from New Mexico State University and is retired from an engineering position at White Sands Missile Range. In 1957 he started Testronic Development Laboratory (now TDL Technology, Inc.) to do consulting and electronic product development. He is still the TDL president and keeps busy in "retirement" doing consulting and technical writing.

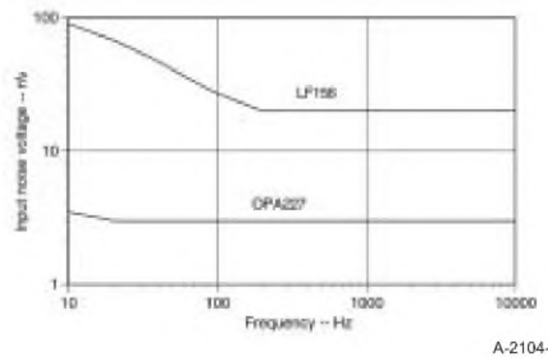

FIGURE 1: Input noise versus frequency for two op amps. In addition to having a lower noise voltage density, the OPA227 has a much lower corner frequency.

PHOTO 1: The completed preamp module.

TABLE 1
1KHZ INPUT NOISE

AMPLIFIER TYPE	MANUFACTURER	NANOVOLTS PER ROOT HZ AT 1KHZ
SSM2017*	Analog Devices	0.95
OPA227	Texas Instruments	3
LM833	National Semiconductor	4.5
LF156	National Semiconductor	20
LM411	National Semiconductor	25
LMC6482	National Semiconductor	37

*Discontinued

I had already decided on a fully metal enclosure with internal rechargeable batteries to minimize unwanted signal pickup. I could regulate the ±9V from the batteries to working voltages of ±5V. After a few rough calculations I had starting resistor values that I could use in a SPICE[3] model to optimize my design.

The SPICE program steps resistor values to minimize the output noise. The final circuit values are shown in *Fig. 2* and in the parts list (*Table 2*). The SPICE output file lists the 2N5087 collector current as 400µA and the 2N5088 collector current as 360µA. (This design may not be truly optimum, but real component tolerances make it impractical to tweak out the last few percent!)

I also decided to build the same preamp with an OPA227 input stage, which I call a model 401B, to see whether the discrete transistor design effort had been worthwhile (*Fig. 3*). I call the discrete transistor version a model 401A, because this makes the test results easy to label.

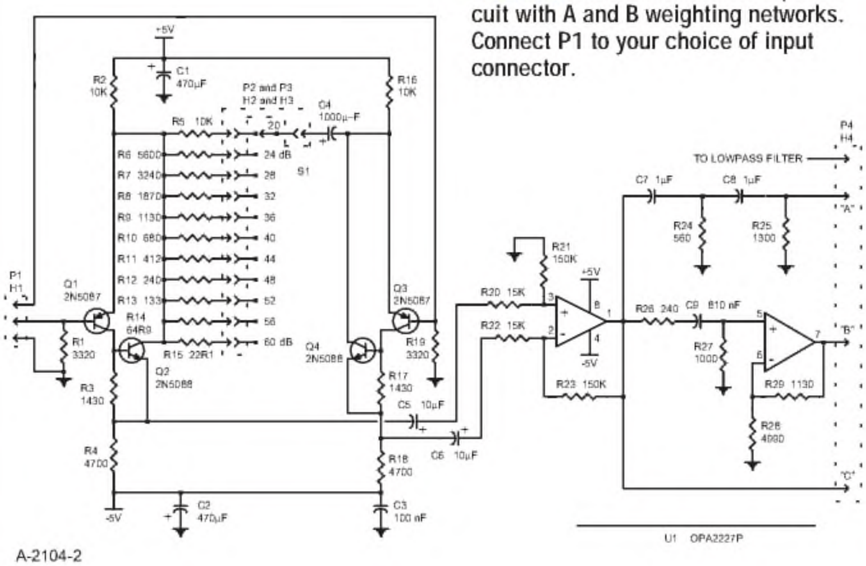

FIGURE 2: Discrete transistor input circuit with A and B weighting networks. Connect P1 to your choice of input connector.

THE REST OF THE DESIGN

I chose a gain range of 10 to 1000 (20 to 60dB) in 4dB steps because 11-position rotary switches are common and because I seldom need a gain of less than ten. I included "A" and "B" frequency weighting filters as well as a flat ("C") response because they are sometimes useful and their addition didn't cost much. The networks use only a few components and the measured response curves are within limits (*Fig. 4*).

The output stage in *Fig. 5* is a 4-pole Butterworth low-pass filter with a cutoff frequency of 25kHz. This reduces the high-frequency noise from the previous stages and may help reduce the noise **aliased** into the passband when driving a sound card's line-input.

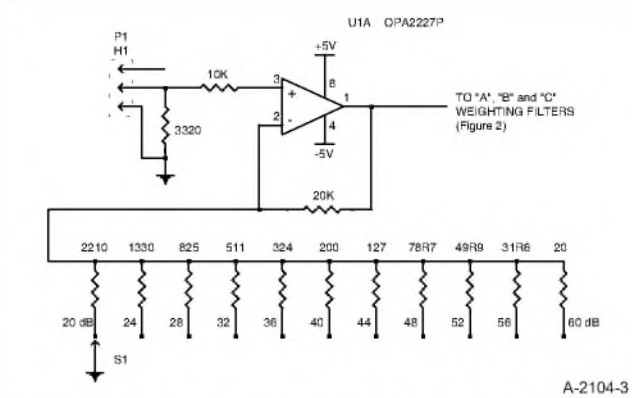

FIGURE 3: Alternate input stage using a low noise op amp.

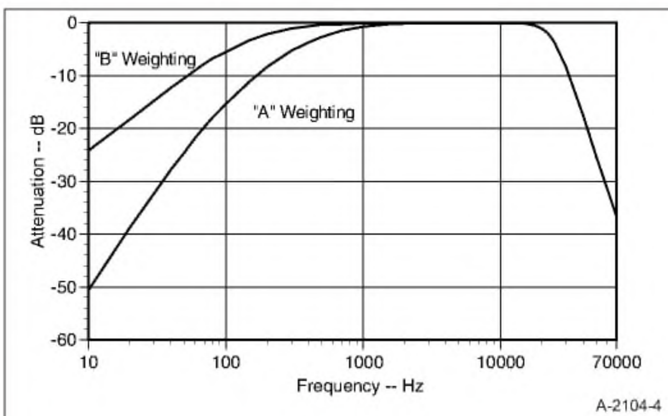

FIGURE 4: Measured response of the "A" and "B" weighting filters.

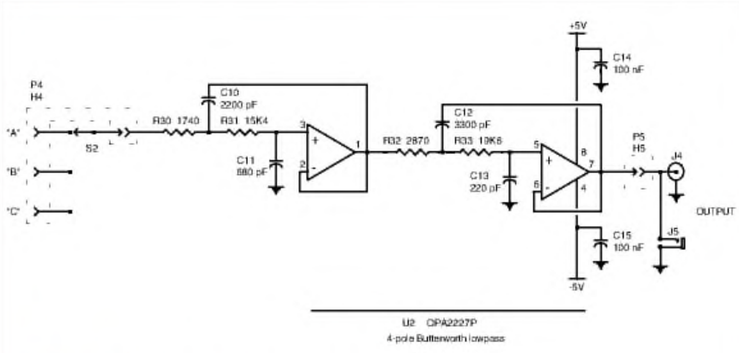

FIGURE 5: Output stage is a low-pass filter.

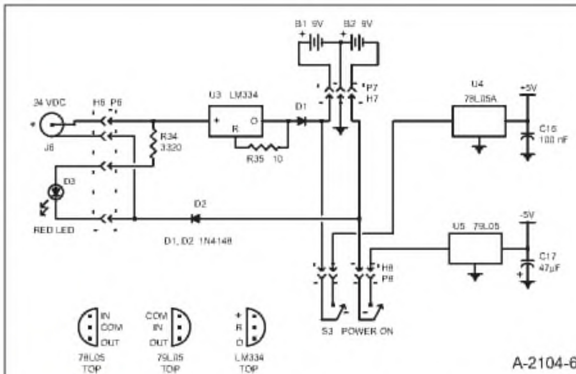

FIGURE 6: Battery charger and power-supply regulators.

TABLE 2
MODEL 401 MIKE PREAMP PARTS LIST

REFERENCE	VALUE
R1, R19	3320
R2, R5, R16	10k
R3, R17	1430
R4, R18	4700
R6	5620
R7	3240
R8	1870
R9, R29	1130
R10	681
R11	412
R12, R26	240
R13	133
R14	64R9
R15	22R1
R20, R22	15k
R21, R23	150k
R24	562
R25	1300
R27	1000
R28	4990
R30	1740
R31	15k4
R32	2870
R33	19k6
R34	3320 (½W)
R35	10

All resistors are 1%, ¼W, metal film, except where noted.

REFERENCE	VALUE	DESCRIPTION	MANUFACTURER
C1, C2	470µF	25V, radial electrolytic	
C3, C14, C15, C16	100nF	50V, ceramic	
C4	1000µF	20V, radial electrolytic	
C5, C6	10µF	25V, radial electrolytic	
C7, C8	1µF	5%, 50V, polyester film	
C9	820nF	5%, 50V, polyester film	
C10	2200pF	5%, 50V, polyester film selected to ±2%	
C11	680pF	5%, NPO ceramic selected to ±2%	
C12	3300pF	5%, 50V, polyester film selected to ±2%	
C13	220pF	5%, NPO ceramic selected to ±2%	
C17	47µF	25V, radial electrolytic	
Q1, Q3	Q2N5087	Low-noise PNP	
Q2, Q4	Q2N5088	Low-noise NPN	
U1, U2	OPA2227P	Dual op amp, 8-pin DIP	
U3	LM334	Constant-current regulator	
U4	78L05A	+5V regulator	
U5	79L05A	–5V regulator	
D1, D2	D1N4148	Silicon diode, 75 PIV	
D3	Red LED	Power-on indicator	Lumax SSI- LXR1612ID (DigiKey 67-1147)
J1, J4		Female, panel mount BNC connector	
J2, J5		Mono phone jack, 3.5mm	
J3		3-pin, male, panel mount audio connector	Mouser 568-NC3MD-L-1
J6		2.5mm male, insulated, panel mount, power input connector	DGS (Mouser 163-4303)
P1, P7		3-pin shell with terminal pins	Molex WM2012
P2, P3		6-pin shell with terminal pins	Molex WM2015
P4, P6, P8		4-pin shell with terminal pins	Molex WM2013
P5		2-pin shell with terminal pins	Molex WM2011
H1, H7		3-pin male header	Molex WM4001
H2, H3		6-pin male header	Molex WM4004
H4, H6, H8		4-pin male header	Molex WM4002
H5		2-pin male header	Molex WM4000
		Terminal pins for the Molex shells	Molex WM2200
S1		Rotary switch, 1-pole, 11-position, make-before-break	Mouser 105-13571
S2		Rotary switch, 1-pole, 3-position, make-before-break	Mouser 105-13571
S3		DPDT miniature toggle switch	
B1, B2		9V rechargeable NiCd	Mouser 573-15F8K
		Cast aluminum enclosure	Eagle type 4593
		Battery holder	Keystone type 1295
		Stick-on rubber feet	DigiKey SJ5523-0
		Knob for rotary switch	
		Circuit board, MAIN401A	
		Misc. hardware, wire and shrink tubing	
		Wall DC power supply to charge batteries, 24V DC @ 100mA	Mouser 412-124013

Figure 6 shows the power-supply circuit, which consists of a 10mA constant-current battery charger and the ±5V regulators. This current is low enough so you can leave the batteries on trickle charge all the time when the preamp is not in use. You can, of course, use the preamp with the charger connected, but you may get better isolation with it disconnected. The battery current is so low the preamp will run for many hours on a full charge.

CONSTRUCTION

Construction is straightforward. As shown in *Photos 1*, *2*, *3*, and *4*, all components except the batteries, connectors, and switches are on the single-sided printed circuit board (*Figs. 7* and *8*). I used three different input connectors and two output connectors because I find it convenient to not have to search for adapters. Switch and connector placement is not critical, and you can use the photos as a drilling guide. The cast enclosure is easy to drill because the hard aluminum alloy cuts cleanly.

All connections to and from the circuit board are made with Molex male headers on the board and mating plugs. I mounted the board on the bottom surface of the enclosure using four 4-40 ×

PHOTO 2: The preamp with the cover removed. Each battery holder is fastened to the enclosure with two ⅛″ aluminum pop rivets.

PHOTO 3: End view showing the three input connectors.

PHOTO 4: End view showing the two output connectors, battery charge input connector, and charge indicator LED.

5/8″ machine screws, four internally threaded 3/8″ long nylon spacers, and four 4-40 hex nuts. Actually, only one or two nuts are needed, because the wiring helps hold the board in place. When the top cover is attached with its six screws, the result is a compact, well-shielded module.

PERFORMANCE TESTING

When testing an amplifier with a sound card, it's useful to put an attenuator between the signal source and the amplifier's input so you have an overall gain of unity. You still have the same amplifier input voltage, but the higher level into the attenuator is much easier to set.

The spectral graphs in *Figs. 9–13* were made with shareware program AtSpecPro[4] and checked with a Hewlett-Packard Wave Analyzer model 3581A. This analyzer has a 90dB single-scale range and a total dynamic range of over 200dB, which is sufficient for testing this preamp.

Although very low distortion audio signal generators are available, they are expensive compared to using software and a sound card. I used a freeware program named Wavetools[5] to generate the tones for the intermodulation distortion tests. Just start the signal generator program twice and set the frequencies and output levels as needed. If you have enough computer memory, you can start this program multiple times to produce a wide spectrum of tones.

Figure 9 shows the model 401A output spectrum at 60dB gain and flat or "C" weighting. The 1kHz output amplitude is just below the maximum sound card line-in level, and you can see that the usable dynamic range is nearly 70dB. The only harmonic visible is the second, and it is down 66dB, which is a THD = 0.05%. (The H-P Wave Analyzer read 0.032%.) The small 60Hz "line" would disappear if either "A" or "B" weighting were used; the flat response is worst case at low frequencies.

As the preamp gain is lowered, the noise floor goes down and the dynamic range increases. The floor is down to about −96dB at 40dB gain (*Fig. 10*), and it's down to less than −100dB at minimum gain

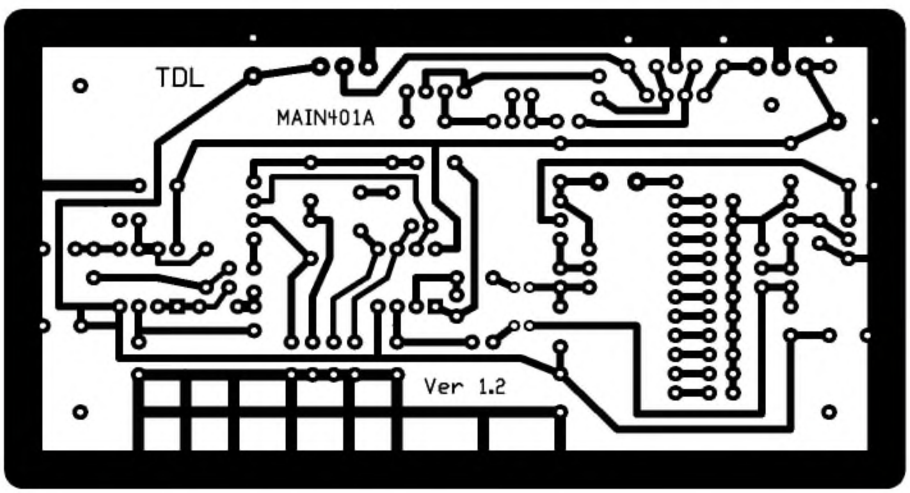

FIGURE 7: PCB (full size). A-2104-7

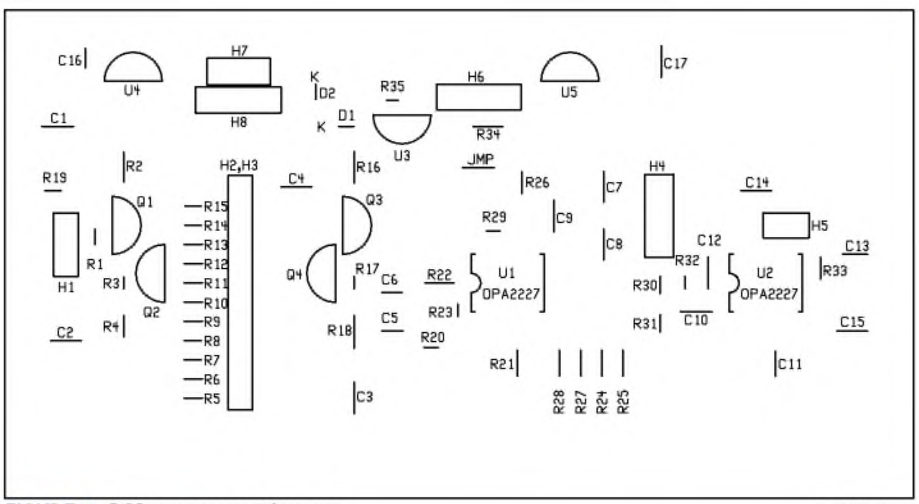

FIGURE 8: PCB component layout. A-2104-8

Circuit boards will be available from FAR Circuits, 18N640 Field Court, Dundee, IL 60118-9269; 847-836-9148; www.cl.ais.net/farcir. The boards are $7.50 each plus $1.50 shipping for up to three boards. VISA and MasterCard accepted with a $3 service charge. Solder plated, single-sided, 1/16″ thick, FR-4 board.

REFERENCES

1. "Sample Champion Pro" version 2.5, by Paolo Guidorzi for Microsoft Windows 95 or higher. You can download a demo: www.purebits.com/download/sctrial.exe (4.2 MB).

2. 2N5087 data sheet from ON Semiconductor. Download the pdf file from http://onsemi.com.

3. Simulation Program with Integrated Circuit Emphasis (SPICE). I use TopSPICE Plus from Penzar Development, but any of the commercial products should work fine.

4. You can download "AtSpecPro" version 2.2 by Paavo Jumppanen from www.taquis.com. The unregistered version has somewhat limited performance but will give you a feel for its capabilities.

5. Download "Wavetools" from www.mda-vst.com. It's freeware!

6. Look for design updates at www.zianet.com/tdl/magarts.htm.

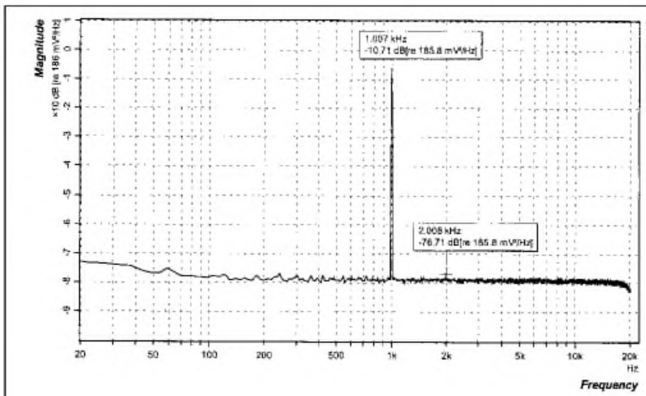

FIGURE 9: Model 401A output power spectrum. 1kHz sine wave input. 60dB gain and flat weighting. Blackman window, 4096 FFT length, and 16 block averaging. A-2104-9

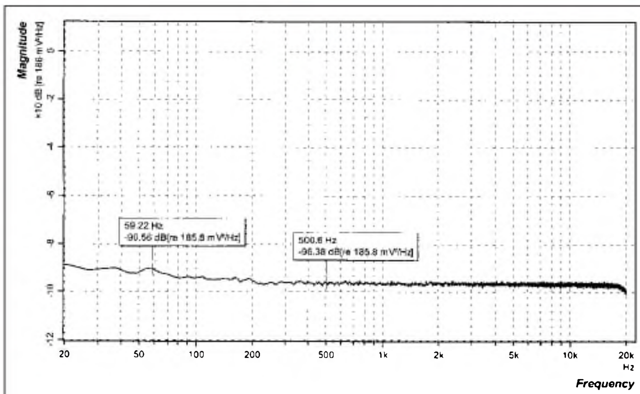

FIGURE 10: Model 401A output power spectrum. No input. 40dB gain and flat weighting. Blackman window, 4096 FFT length, and 16 block averaging. A-2104-10

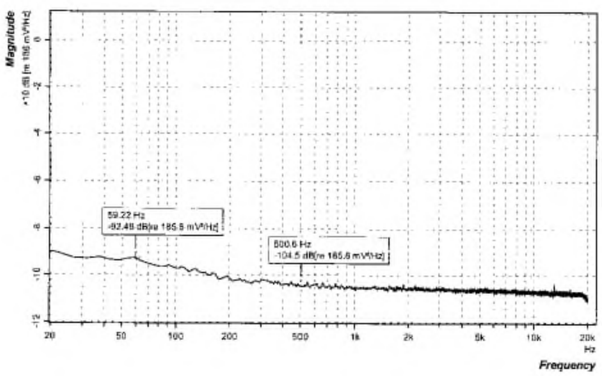

FIGURE 11: Model 401A output power spectrum. No input. 20dB gain and flat weighting. Blackman window, 4096 FFT length, and 16 block averaging. This looks the same as the sound card's noise level. A-2104-11

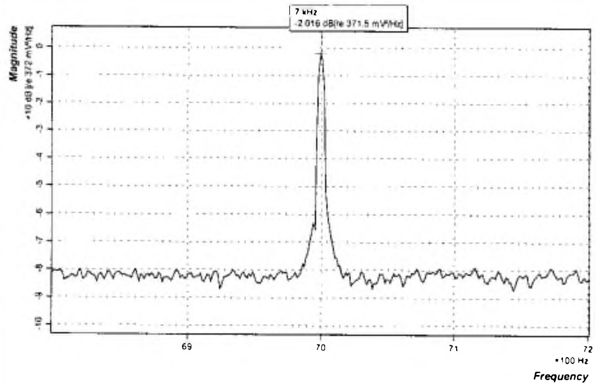

FIGURE 13: Model 401A output power spectrum for intermodulation distortion test. 60dB gain and flat weighting. Blackman window, 16384 FFT length, and 16 block averaging. A-2104-13

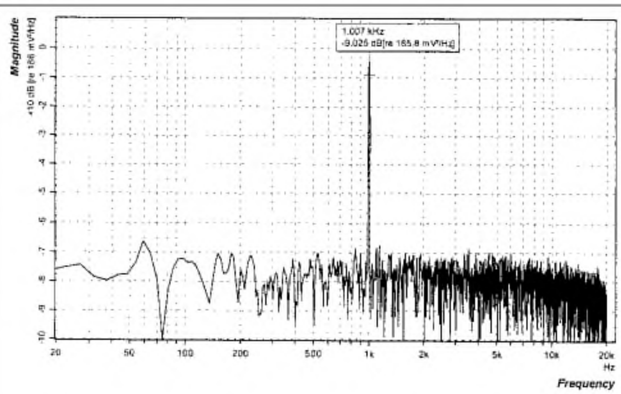

FIGURE 12: Model 401B output power spectrum. 1kHz sine wave input. 60dB gain and flat weighting. Blackman window, 4096 FFT length, and 16 block averaging. A-2104-12

(*Fig. 11*). This is at the sound card's limit, and a spectral graph made with the sound card's input shorted looks the same as *Fig. 11.*

In *Fig. 12* you can see the model 401B response for the same test conditions. The noise floor is about 10dB higher so the dynamic range is 10dB lower. Clearly the discrete transistor input performs better.

Figure 13 shows the output spectrum of an intermodulation distortion test on the model 401A. I used a 7kHz sine wave with a relative amplitude of unity summed with a 90Hz sine wave with an amplitude of 0.1 as the input. No IM products are visible. Nor were they visible for the model 401B, but the noise floor was higher.

So how does it sound? Well, it's very quiet. I connected a Sony condenser mike to the preamp's input and the preamp output to the input of one of my sound systems. With the mike off, the sound system gain at maximum, and the preamp gain at 44dB or lower, I couldn't hear anything out of the speakers. At 48dB I heard a faint hiss if I turned off the fluorescent lighting. At 60dB the hiss is barely audible. Record-

(to page 69)

Creative Uses for "A Universal Digital Out"

Expand the flexibility and usefulness of your audio equipment with digital output. **By Ali Elam**

Dale A. Beard's article, "A Universal CD-Player Digital Output" (*AE* 1/98, 2/98), was highly informative, and the project was neat. I was looking for an excuse to build one but that turned out to be a challenge. After all, even today's lowliest CD players have digital outputs. As luck would have it, though, just as I started to forget about it, opportunity knocked—in fact twice!

SEARCH FOR DIGITAL OUTPUT

I recently acquired a pair of Philips DSS930 digital speakers. These have only a coaxial digital input, and carry out the crossover function—as well as equalization and anticipative phase correction—in the digital domain. The output is then put through a pair of DACs and biamplified to drive a tweeter and a pair of midwoofers. The result is that the frequency response is ruler-flat and the phase is well-behaved.

But, they only take a digital input. The problems produced with analog inputs (tuners, cassette players) are self-explanatory. Not immediately obvious however, are the issues with the play-

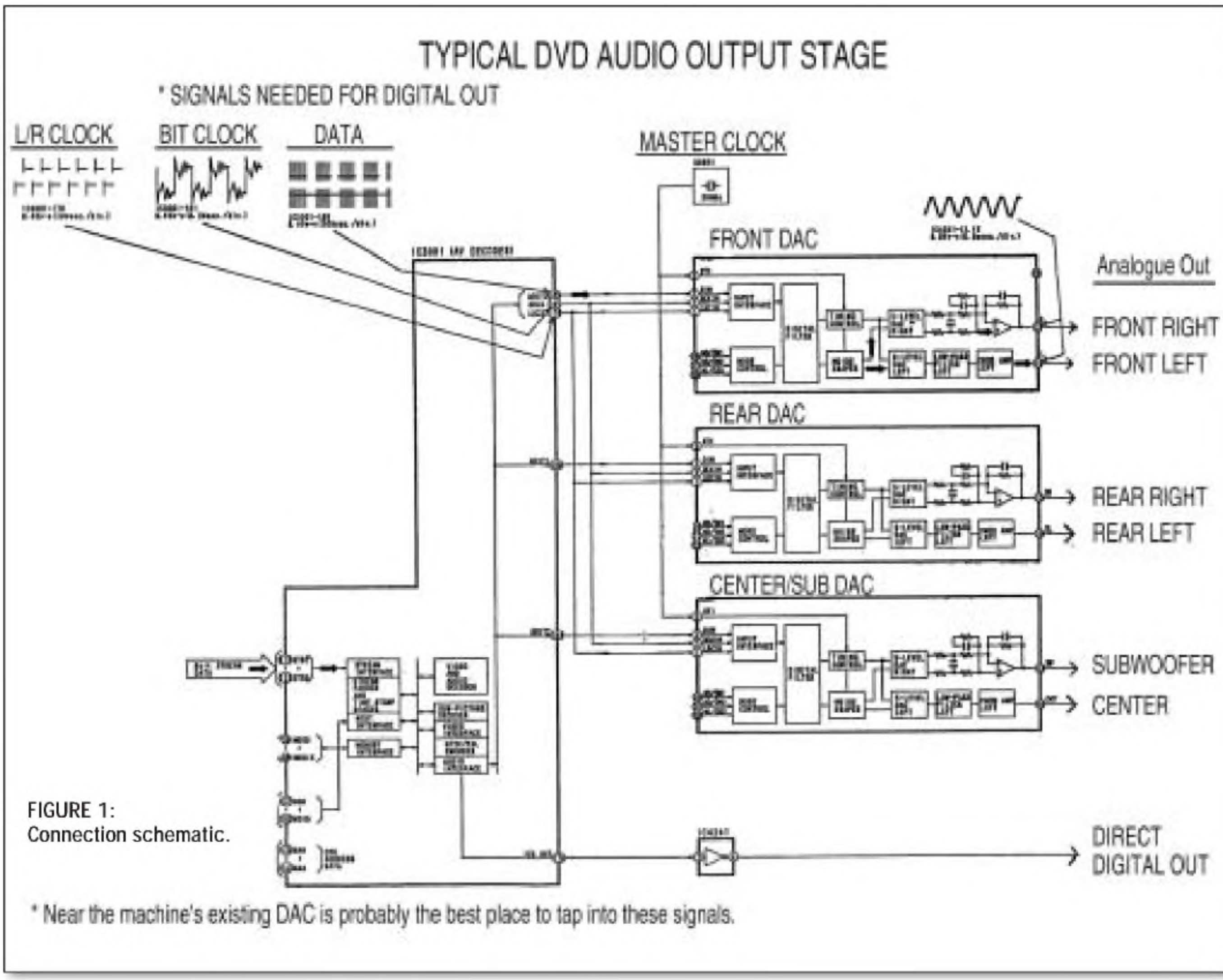

FIGURE 1: Connection schematic.

back of DVDs (which are also eponymously digital). Even if your DVD player has a digital out, read on before you burn your speakers.

The digital outs on DVD players output whatever is on the disc–PCM (such as CDs), Dolby AC-2, AC-3, and DTS. So, when you play a CD or a DVD with a PCM soundtrack (some DVDs give you a choice), the digital out is just like a CD player's. If you play anything else, the output is a coded signal, which through a "normal" DAC sounds like a high-pitched (and extremely loud) warble.[1]

I wished to drive my digital speakers directly or make CD/MD copies of my music DVDs to play in the car or in a regular CD player. So, I started looking for a Dolby Digital decoder with digital outs in the PCM mode. Frankly, even thinking about this is puzzling–a normal PCM digital out contains two channels: right and left. How do you pair Dolby Digital's six outputs: R(f), L(f), center, R(r), L(r), and subwoofer?

My research resulted in only one such decoder—the Meridian 568—which is designed to drive their digital loudspeakers. At $6500 (although it did many other things) I decided to give it a pass. I had several conversations with the folks at Meridian and e-mails with Dolby (both were extremely helpful and patient with my nosing about). I also ordered the service manual for my DVD player, which has a built-in Dolby Digital decoder (as many do these days).

A SURPRISING DISCOVERY

What I found out was fascinating! The audio visual (AV) decoder (a massive LSI) has four sets of digital outputs: one is the "straight through" of whatever's on the disc that goes to the machine's digital out (*Fig. 1*). The other three are paired as follows:

- Front channels, right and left
- Back channels, right and left
- Subwoofer and center channel

The first two pairings are what you would expect, and I thought the last was imaginative and somewhat funky (a binary stream that alternately contains Godzilla's thump or Garbo's murmuring). These sets (with their clock, signal, and other combinations) drive three DAC chips for the analog output.

I guess I could stop here, because the detective work has been accomplished. You simply need to build up to three digital-output circuits, connect them to the signal and clock signals, and label them correctly (in the last pairing the "normal" right is the subwoofer and the left is the center channel).

I say "up to three" because the number of these signals you will want to have access to in the digital domain depends on their intended use. If you plan to make CD/MD/DAT copies of music DVDs, just the front channels will suffice. In fact, most of my opera and rock DVDs are in the AC-2 format, which is essentially a compressed two-channel scheme. Today's music DVDs cost only a couple of dollars more than CDs, so whenever available I opt for them. But, it is convenient to be able to make copies playable just as music CDs.

There are any number of reasons to go the full-blown, six-channel route. Many of you believe that the DACs in your CD players are inadequate and you should upgrade them with external units. Here is your chance to do the same for the DVD player–except with triple the fun, and you can put all your ex-DACs to use again. Maybe a nonoversampling DAC will sound great on the subwoofer channel. Alternatively, you may get lucky and find a set of digital Philips or Meridians all around[2]. I built three output circuits myself, but ended up using only one in the DVD player because other opportunities beckoned.

OTHER USES

My next digital output went into an analog cassette deck. This sounds oxymoronic, but Pioneer has a newish (1997) deck called the "W606DR." It has full DSP inside, which, among other functions, also performs noise reduction. According to their literature, this is done by spectrum-analyzing the music and eliminating uncorrelated, random noise. This sounds gimmicky, but works uncannily well. I suppose it is not equivalent to CEDAR de-hissing, but at $200 for the entire deck, it works flawlessly. Even nonDolby tapes get a CD-like silence, and to my ears, it is transparent for music[3].

I thought that I could make CD-R copies of my old tapes on their "final journey." Again, just like with the DVD, why take the deck's analog output and convert it to digital, when you know somewhere in the DSP it is already digital? Scouring the ever-handy service manual (over several nights) again provided the clues.

IN SUMMARY...

I actually never built Dale Beard's circuit. It looks like the "Cadillac" of digital outs, with onboard regulators and reset circuits. As in real life, I went with the "Saab" instead: Erik Gustavsson's Swedish site (www.algonet.se/~cyrano/unispdif.html) has a minimalist circuit that uses the same Crystal Semiconductors chip[4]. It has the PC layout and instructions and is very complete.

In my experience, there are only a few tricky bits to this project: You need to find six leads in the host machine:

ABOUT THE AUTHOR

Ali Elam is a senior vice president at Lehman Brothers, the investment bank, in New York City, where he also lives with his wife and two young sons. He holds an engineering degree from Yale and has been a life-long electronics hobbyist. In addition to a "passing" interest in digital, he collects and restores antique reel-to-reel tape recorders and black and white TVs.

REFERENCES

1. Interestingly, a "bit perfect" CD recorder will copy this signal without "understanding" what it is, and a CD player will play the copy back, which can be fed through its digital out to a Dolby/DTS decoder and replayed.

2. The Philips DSS930s that I bought came out in 1992 and retailed at £4000 (about $8000 at the time). I remember one of the British hi-fi magazines raving about them but saying, "not at the price..." I was walking on NYC's Canal Street six months ago and saw three of them in a surplus store. They were allegedly "factory close-outs" (I never knew they were sold in the US). One was lacking a woofer, another the amplifier circuit board and the tweeter. I bought the lot for $100. None of them worked. It turned out one had a blown power transformer and the other a bad ground. Another $200 got me a new tweeter, a remote (that is the only way you can turn them on or adjust the volume), and some cosmetic parts—all special-order from Holland. A few nights of veneering, polishing, and painting made them much better than new (the originals were clad in vinyl).

3. The British *HFN&RR* magazine was equally impressed. Personally I was so taken by it that I pulled out leads from the playback amp to run my reel-to-reels through the DSP. With some level setting, that also works well.

4. I bought all my parts except the Crystal chip and the optical output LED, from Debco Electronics. Those came from SC Electronics in Manchester UK. My cost per unit was about $30, and half of that went to the Crystal chip.

5. Unfortunately the 8402 chip used in this project does not accept all the data formats used in today's machine (such as "LSB justified 20 bits").

1. The master clock (MCK)
2. The bitclock (BCLK)
3. The left/right clock (SLRCK)
4. The serial digital data (DIN)
5. The +5V, and
6. Ground.

Unfortunately, each hi-fi manufacturer seems to use different acronyms for the first four. It is best to look up on the internet the DAC chip the component uses and try to figure out from its manufacturer's spec sheets which pins refer to the required signals because these sheets have text explanations. The master clock is the trickiest because it shows up under unrelated names such as XTI as well as the more common MCK. Not all clocks have the same frequency as the master—they may be only half the master, for example. But that is "half" the fun.

The other tricky bit is finding the clock frequency and data format. As there are only a handful of selections (entered on Gustavsson's circuit with a dip switch), trial-and-error is possible.[5] ❖

The DR12a Horn

If you're looking for a home for your 12″ driver, try this cabinet, which is small in size, but big in performance.

By Bill Fitzmaurice

PHOTO 1: The DR12a.

I'm going to preface this project with a bit of advice that you generally won't find most authors giving: don't build this speaker. It is not the best choice for its intended application, and you would be better off, all things considered, to use a different design.

That being said, let me clarify. I designed the DR12a for those of you who want a DR live-sound bass/PA/keyboard cabinet with a 12″ driver, but don't need the output capability of the 80 lb, 13ft^3 DR12. By contrast, the DR12a is only 8ft^3, and at about 60 lbs when loaded with a premium driver is not too much for one person to handle. Its performance exceeds any commercial single 12 (or single 15, for that matter) box by a wide margin, and is second only to the DR12 in sensitivity and bandwidth. If you must use a 12″ driver and want the highest possible sensitivity and bandwidth from the smallest possible cabinet, the DR12a is it (*Photo 1*).

On the other hand, from a cabinet of equal size–my DR10–you can get a better result using a 10″ driver. Compare the response of the two cabinets in *Fig. 1*. The DR12a, which uses a 100dB EV12L™ driver, has somewhat higher sensitivity. If the tested DR10 were loaded with a 100dB driver instead of its 98dB Eminence™, it would be higher in sensitivity than the DR12a. Conversely, if loaded with a midline driver, the DR12a would be less sensitive than the DR10.

That a ten can out-power a 12 in cabinets of similar size is the product of the more efficient horn section in the DR10, made possible by the smaller frame size of the 10″ driver. That smaller frame also makes construction easier, gives more clearance inside the cabinet for ease of mounting the driver, and lowers cost (a ten is cheaper than a 12). If you want the best 8ft^3 box out there and are starting from scratch, the DR10 is the way to go. But if you already have a 12 or two on hand and don't want to shell out for more drivers, the DR12a is the next best thing.

DESIGN FEATURES

Like all the DR (Double Reverse) folded horns, the 12a is two horns in one. The driver feeds into a short, rapid flare midrange horn, which in turn feeds into a pair of longer, slow flare bass horns. The connection of the horns, where the sound path reverses, is a bend formed by nearly concentric arcs. This is a departure from traditional folded horns, which use flat reflectors in their horn bends.

When sound waves bounce off flat surfaces out-of-phase reflections occur, especially at high frequencies. The result is poor high-frequency response; most folded horns roll off high frequencies at 500Hz or less. As you can see in

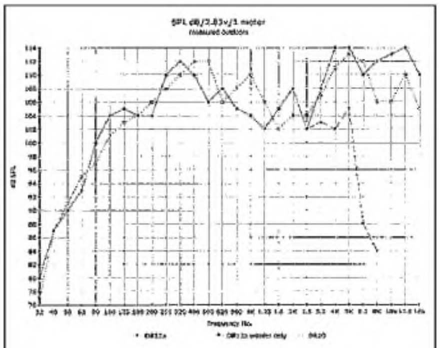

FIGURE 1: SPL comparison—the DR12a with EV premium driver versus the DR10 with midline Eminence driver. B-2138-1

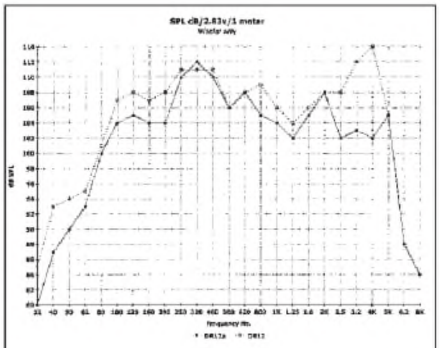

FIGURE 2: SPL comparison—the DR12a versus the larger DR12, loaded with the same EVM-12L driver. B-2138-2

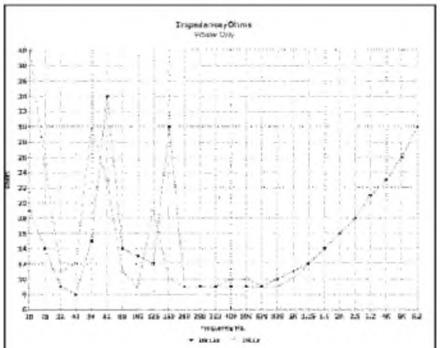

FIGURE 3: Impedance, DR12 and DR12a. B-2138-3

Fig. 2, the woofer in the DR12a is usable up to 5kHz.

Other noteworthy points are evident in *Figs. 2* and *3*, which compare the DR12a to the larger DR12. First, I have read that below a horn's flare frequency a horn will not pass sound. Some have even suggested that at frequencies where the horn length is less than ¼ wavelength, sound waves will never leave the horn, falling backward into a sort of sonic "black hole." To this I can say: 't ain't so!

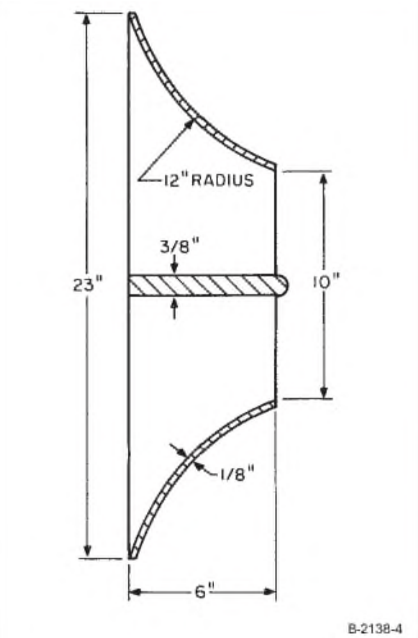

FIGURE 4: Side view of the throat horn.

Impedance lows show that the DR12 has an Fh (horn frequency) of 100Hz, while the smaller DR12a Fh occurs at 125Hz. Neither shows any "brick wall" effect below Fh, both rolling off at 12dB/octave until port output begins to dominate.

The larger horn is more efficient than the smaller one at almost all frequencies, not just those below the Fh of the smaller horn. These horns have a similar design, were tested with the same driver, and the only noteworthy difference between the two that might account for this sensitivity difference is the size of the mouth openings. This meshes with my theory that increasing radiation area increases efficiency, whether by using multiple drivers in a direct radiator or by increasing mouth size in a horn.

A crucial design aspect arises from the differing Fs(h) of the two cabinets. [Fs(h) is the resonant frequency of the driver/horn combination.] For best performance the Fh/Fs(h) ratio should be from 2:1 to 3:1. The 100Hz Fh of the DR12 allows a 32Hz Fs(h), which gives the box superior lows.

The DR12a Fh of 125Hz forces raising the Fs(h) to prevent a response dip in the second octave. The problem here is that in order to raise the Fs(h) of the DR12a its throat must be increased to 63 in^2, compared to 51 in^2 in the DR12. Increasing throat size raises Fs(h), but it also hurts high-frequency loading, the result of which you can see in *Fig. 2*.

This is a trade-off you must accept in order to have a smaller cabinet. Luckily, once the DR12a's tweeters are added, the highs are restored, though DR12a still does not approach the high-frequency excellence of the DR12.

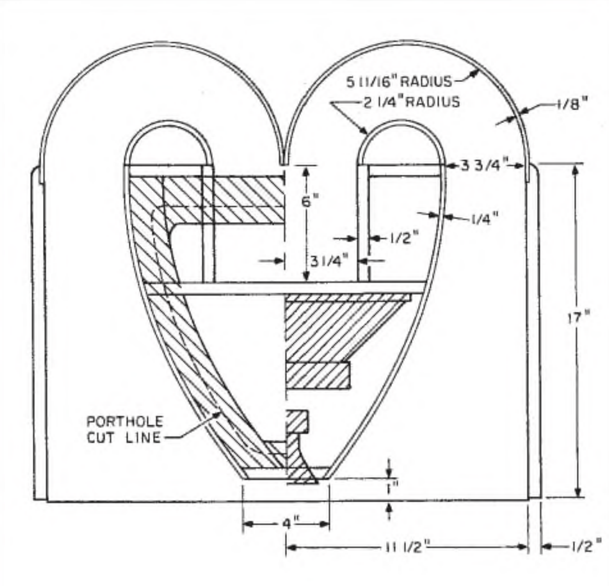

FIGURE 5: Top/bottom view. Half of view shows interior parts, half shows the horn braces (shaded) and port hole cut line.

DRIVER SELECTION

The Fs(h) dilemma dictates the choice of drivers for this project. The recommended EVM-12L™ has a specified Fs of 55Hz for an Fs(h) of 45Hz (my sample measures at 48Hz, giving an Fs(h) of 38Hz). You could go as low as a 40Hz Fs, which would give an Fs(h) of about 32Hz. This would provide a bit better response from 32Hz to 40Hz, but would probably cost 3dB at 80Hz. I wouldn't go that route unless I were using the box for either five string or keyboard bass.

At the other end of the scale, a driver Fs of 60Hz would give an Fs(h) of about 50Hz. This would add perhaps 2dB from 80 to 100Hz, but you'd lose that much below 50Hz. The best Fs is one as close as possible to 55Hz.

Although I built the prototype to accept an EVM™, other premium drivers also should fit. On the other hand, a premium driver is not required. Even a stamped frame MI driver should work well, provided the Fs is on target, and the rated SPL of the driver is at least 97dB. As for V_{AS} and Q_{TS}, tests of a DR horn with three different drivers has led me to believe that V_{AS} and Q_{TS} matter little in these cabinets, so I am no longer considering those specs.

The tweeters on the prototype are Motorola™ Twin-Bullets, model KSN1177. With two piezo elements apiece, a pair of these is very potent. If you opt for a different model, you may use from two to four elements with good results. Two elements will give a fairly flat response, while four will give ultra-sensitivity in the high end for maximum cutting power with minimum distortion.

For electric bass you may omit the tweeters, but a good reason to have them is so your stage sound will be the same as the out-front sound through a full-range PA. If you intend your DR12a for PA use, you may want to add side-fire tweeters as well (see DR10 (Jan. '01) or DR8 (Nov. '01) projects for details).

MATERIALS

Taking advantage of the self-bracing design, I kept the materials used for the prototype as thin as possible. For ease of joinery you may use thicker materials, adjusting dimensions accordingly. Slight alterations to the cabinet size won't affect performance significantly, though making the driver chamber smaller may prevent you from getting the woofer into place.

Use a decent grade of plywood, but don't waste money on premium materials. I used a five-ply ½″ underlayment plywood for the top, bottom, sides, and other flat parts. The back sheaths should be made of ⅛″ Baltic birch if available, though ⅛″ lauan will do in a pinch.

The mouth horn panels can be either ¼″ or ⅜″ plywood, but don't use Baltic there, because it is too stiff. Use spruce plywood with three equally thick plies, which bends very easily. A poor choice here is ¼″ lauan, which usually has a thick center ply faced with two thinner ones and doesn't bend well. Before laying out any bending parts to be cut, flex the plywood to determine its easier bending axis.

The throat horn braces, back braces, and horn braces may be as thin as ¼″ to save weight (the prototype specifies ⅜″), but that dimension won't hold screws. Using ½″ plywood for those parts will simplify construction by allowing use of screws, at the cost of additional weight. And while the ⅛″ thick back sheaths will not vibrate, they also won't hold up to a serious bashing by a careless roadie, so you may want to laminate a second layer there if you don't load your own gear.

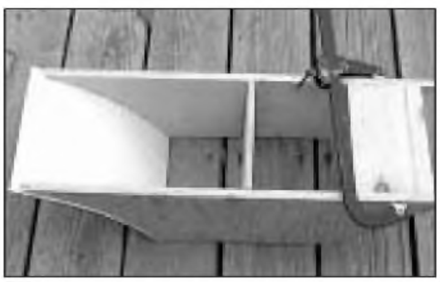

PHOTO 2: Using clamps and a temporary brace while assembling the throat horn.

TABLE 1
PARTS LIST

Throat horn sides 23″ × 6″, ½″ plywood
Throat divider 6½″ × 6½″, ⅜″ plywood
Throat horn sheaths 10″ × 7½″, ⅛,″ or ¼″ plywood
Throat horn supports 23″ × 3¾″, ½″ plywood
Top, bottom 23″ × 23″, ½″ plywood
Tweeter baffle 23″ × 4″, ½″ plywood
Baffle 13½″ × 18″, ½″ plywood
Woofer spacer 12½″ × 12½″, ⅛″ plywood
Horn braces 3″ × 17″, ½″ plywood
Mouth horn panels 23″ × 18″, ¼″ or ⅜″ plywood
Horn bend reflectors 2′ × 4″ Schedule 40 PVC
Back braces 6″ × 11½″, ⅜″ plywood
Side braces 10″ × 15″, ⅜″ plywood
Back sheaths 20″ × 24½″, ⅛″ plywood
Ducts 2′ × 1¼″ PVC electrical conduit
Sides 17″ × 24″, ½″ plywood
Parts are listed with nominal dimensions; actual dimensions will vary with true thickness of materials used and must be verified by dead reckoning during the construction process. Thicker materials than those specified may be used for non-bending parts. Not all parts are listed—see text for details.

The ducts are fashioned from 1¼″ PVC electrical conduit, which is stiffer than regular PVC pipe and is also dark gray in color, eliminating the need to paint them. The horn bend reflectors are fashioned from 4″ Schedule 40 PVC pipe. Both the conduit and the pipe may be cut with a regular saw blade, but an abrasive blade works better.

Most joints are fixed with drywall screws and construction adhesive, with screw holes bored and countersunk. You'll also need a hot-melt glue gun where quick setting is required. The rule of thumb is to use construction adhesive where screws will hold the joint tight until it cures, and hot-melt everywhere else.

PHOTO 3: The throat horn with sheath stiffeners in place.

PHOTO 4: Using clamps and guideboard when attaching throat horn support.

PHOTO 5: Trimming and squaring the throat horn assembly.

Wherever possible, clamp parts in proper alignment before drilling holes and applying adhesive. Also note that many of the interior parts have holes cut in them with a hole-saw to minimize internal reflections. This can lop another pound or two off cabinet weight as well, which never hurts.

CONSTRUCTION

Begin construction by cutting out the throat horn sides (*Fig. 4*). Save the trimmings from the radius cuts for later use. The sides are joined to the throat horn divider, offset to one side of the

PHOTO 6: Using a long wooden compass to trace arcs.

PHOTO 7: Using clamps and guideboard attaching throat horn to top.

PHOTO 8: The assembly after attaching the baffle.

centerline so that the two throat horn halves are slightly different in size, with the rounded rear edge of the divider extending a half-inch past the driver-side edges of the sides.

The throat horn sheaths are glued and screwed into place. Use a temporary shim and clamps to hold the throat horn sides in alignment while you attach the sheaths (*Photo 2*). When complete, run the saved trimmings from the sides down the middle of the sheaths to act as stiffeners (*Photo 3*).

Cut the throat horn supports a bit longer than the length required, and drill a few 2″ holes in them. Attach them to the throat horn assembly, using a straight guideboard and clamps to align the parts perfectly before gluing and screwing (*Photo 4*). After the adhesive has set, run the completed assembly across a table saw, using a panel cutting jig, to trim the assembly square and to size (*Photo 5*).

Cut the top and bottom pieces (*Fig. 5*), and draw onto them the locations of the mating parts, and the port hole cut line in the bottom, using a long piece of wood with a screw through one end and a pencil through the other as a large compass to trace arcs (*Photo 6*).

PHOTO 9: Spacing of horn braces.

PHOTO 10: The flanged porthole. Note trimming to allow driver entry.

PHOTO 11: Pulling a mouth panel into place with clamps.

PHOTO 12: Assembly after cutting tweeter and duct mounting holes.

PHOTO 13: Holding PVC pipe with a jig for slicing on table saw.

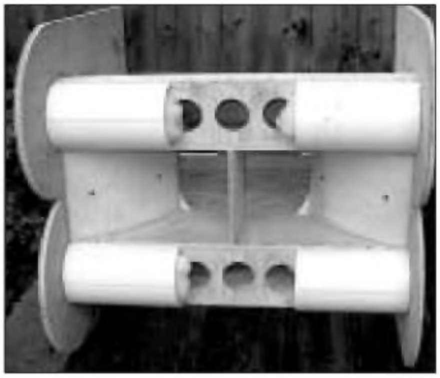

PHOTO 14: Attaching the PVC reflectors; note polyfill stuffing.

Again using a guideboard and clamps, attach the throat horn assembly to the top (*Photo 7*). Cut the tweeter baffle with the edges at a 40° angle; save the trimmings from the edge cuts, being sure to make them at least 1½″ wide.

Attach the tweeter baffle to the top. Cut the port hole from the bottom and attach the bottom to the assembly. Cut the baffle, noting that one end of it attaches to the top, while the other is made only long enough to hold the driver. Trial-fit it against the throat horn, tracing the cut out on it from within the throat, and cut the baffle opening. Fashion a shim from thin plywood to space the driver away from the baffle, cutting a 10½″ hole in the shim. Attach the shim to the baffle.

Drill pilot holes and install four ⁵⁄₁₆″ T-nuts in the baffle for driver attachment. Put the holes at the corners of the baffle only; bolts at the edge will not be reachable anyway once the cabinet is complete. Drill some large holes in the baffle assembly. Use clamps and a T square to pull the assembly into alignment, and then attach the baffle to the assembly (*Photo 8*).

Using the wooden compass technique, make a pattern for the horn braces and duplicate it seven times from plywood or stock lumber. Install two of the braces whole, producing a flange on the port hole opening on the bottom. Install two on the top, bisected to accommodate the baffle, while cutting the other two similarly to fit between the baffle and the horn braces and the tweeter baffle (*Photo 9*). Trial-fit the driver to be sure that the braces won't interfere with either the driver or the mounting bolts. Use scraps to finish flanging the porthole opening, trimming as necessary to allow the driver frame to clear (*Photo 10*).

Trim to fit and attach the tweeter baffle edge selvage strips to the tweeter baffle, to double its thickness. Cut the mouth horn panels and install them, attaching them first at the tweeter baffle, using clamps to pull them to the horn supports (*Photo 11*). When the adhesive has cured, sand the panels flush to the tweeter baffle and horn supports.

Cut mounting holes for the tweeters in their baffle, located so that the installed tweeters will not hit the woofer magnet. Cut a piece of 1¼″ PVC conduit at a 40° angle, using it to trace four oval duct holes on the mouth panels (*Photo 12*). Cut the holes and use a rotary sanding drum to chamfer the holes to accept the ducts.

Using a cutting jig on a table saw (*Photo 13*), halve lengthwise a 2′ piece of 4″ PVC pipe. From this, cut three not-quite identical pairs of reflectors, their combined lengths plus the thickness of the two back braces equaling the total interior height of the cabinet (in the case of the prototype, the pairs measured 7¼, 7½, and 7¾″). Using reflectors of differing lengths randomizes the spacing between the braces and the top and bottom, to minimize phase reflection anomalies.

Chamfer the mating edges of the PVC parts to act as a trough for adhesive. Stuff two sets of the pairs with polyfill and hot-melt them to the horn supports (*Photo 14*). Cut the horn supports and install them to the assembly, and then stuff and install the remaining PVC reflector pair (*Photo 15*). Make sure that all edges are amply glued for an airtight fit.

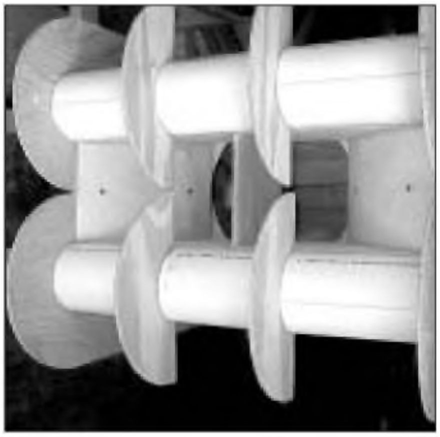

PHOTO 15: Assembly after installation of back supports and last two reflectors.

PHOTO 16: Tracing final cut line onto a side brace.

Cut the horn braces from the same material as the back braces, making them about ½″ oversize. Temporarily clamp them in place, using a pencil to mark on them the true curvature of the horn sheaths, and trim them accordingly (*Photo 16*). Attach the horn braces, driving screws through the sheaths from the cabinet interior if they are thick enough; otherwise, hot-melt them.

Use a straightedge run from the top to the bottom to define the finished outside edge of the braces and trim them. Cut the back sheaths and install them, starting with both sheaths inserted into the back cavity, tacked in place with hot melt (*Photo 17*). Gradually pull the sheaths into place with a webbing clamp and wood cauls, jointing as you go. Extend the sheaths at least an inch past the curvature of the back. Laminate a second layer of back sheathing if so desired. Cut the side pieces and dado them to accommodate where they will overlap the sheaths (*Fig. 6*).

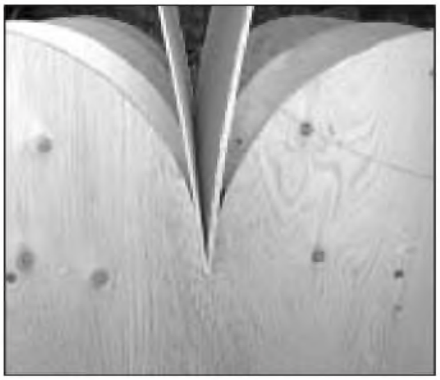

PHOTO 17: Beginning installation of back sheaths.

PHOTO 18: Using a webbing clamp and wooden caul to hold side in place for assembly.

Attach the sides, again using clamps and cauls to hold the assembly together until the adhesives have set (*Photo 18*). Sand all the joints smooth. Any exposed reflecting surfaces within the cabinet will cause response anomalies, so eliminate them wherever possible. Line large surfaces with glued-on acoustic foam, and lightly but thoroughly stuff all voids within the box with polyfill.

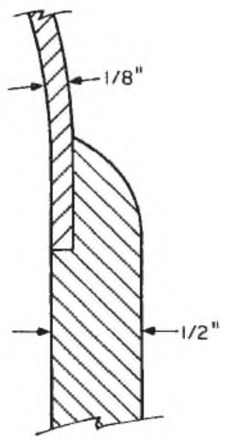

FIGURE 6: View of the dadoed side overlapping the back sheath. B-2138-6

DUCT WORK

The next step involves duct sizing, which requires impedance testing. If you don't have testing equipment, you can wing it by making all the ducts 4″ long, giving an f_B of about 45Hz. This should prove adequate assuming a driver Fs of 55Hz.

Cut the duct ends at 40° to match the taper of the panels, and glue them into place from the cabinet interior, with the outer joint sanded smooth flush to the panels. Remember that the effective length of the ducts does not include the section where it is tapered.

If you do have testing equipment, install the woofer, using Allen-head mounting bolts, leaving the duct, tweeter, and port hole openings clear. Run an impedance plot to determine the Fs(h) of the cabinet, which is represented by an impedance spike about 10Hz below the Fs of the driver. Install three ducts each 6″ long. Run the wire to the woofer out of the cabinet through one of the ducts. Temporarily seal the tweeter openings with gasketed plywood scraps. Gasket the port hole cover and temporarily install it.

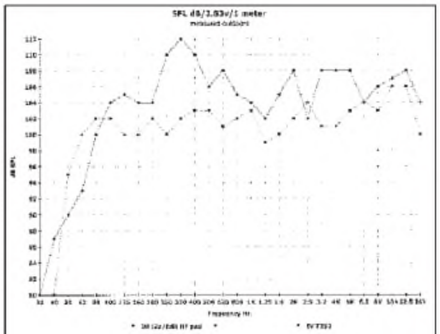

FIGURE 7: SPL comparison, the DR12a versus the EV T252+. B-2138-7

Test the impedance again; you want the f_B, which will appear as an impedance null, to lie at the same frequency as the previously determined Fs(h). This is achieved by altering the length of the final duct; varying it from 6″ to zero will shift Fs(h) from about 35 to 50Hz.

When you've determined the length of the last duct, open up the box and install it. Now apply your finish of choice. Paint is cheap, but looks cheap, too. Plastic laminate is expensive and time consuming to install, but gives a professional look and is durable, especially when combined with aluminum edging. On my cabinets I use carpet, which looks good, is easy to apply, and is reasonably priced.

Install the tweeters, gasketing their flanges (unless you use a carpet finish, which acts as a gasket). For PA use, install side-fire tweeters on the cabinet sides. All tweeters are wired in parallel, in phase with the woofer. Install your jack of choice; you may put it on the port hole cover or on the cabinet back or side as desired. Seal airtight any holes you may drill into the driver chamber for jack or side-fire tweeter wires to pass through.

If you are using the box for PA, you can install a stand-mount top-hat on the port hole cover as well; before you drill a hole for it, determine the cabinet center of gravity for its mounting position, and check that the top-hat won't hit the woofer magnet. Make sure the cabinet stuffing does not cover the entrances to any of the ducts or the rear of the woofer cone. Install the port hole cover, casters, and handles as desired. Enjoy.

DIY VERSUS COMMERCIAL

The original investor who commissioned me to design the DR series of folded horns has opted out of the project for lack of funding, so if any of you out there are looking for a place to invest your lottery winnings, let me know. Meanwhile, I've begun to shop around for a company that might want to build a commercial version of these boxes, but the Not-Invented-Here syndrome has so far stymied my efforts. Too bad, because these horns really outperform so-called "State-of-the-Art" T/S boxes.

As an example, check out *Fig. 7*. Here I've compared the 80 lb/8ft^3 DR12a with the 114 lb/11ft^3 Electro-Voice™ T252+. It is an unfair comparison, because the DR12a has only a single 12, while the T252+ is loaded with two 15s. Unfair to the T252+, that is, because within a T/S box two 15s have no chance against a single 12 in a DR horn. (To make the comparison more useful, note that the chart reflects a 6dB padding of the DR12a tweeter section, as is the case with the EV.)

The list price of the T252+ is $1779; you may find them sold at discount for $1200. Why anyone would pay that kind of money for just another vented box is beyond me, but does help explain why EV™, among others, isn't interested in producing a superior cabinet: cheap-to-produce T/S boxes equal higher profits, and low SPL efficiency means they get to sell more of them.

I guess that's called capitalism, but you don't need to stage a revolution to beat the system. Assuming that you already have a good 12 on hand, you can build the DR12a as described for about $100. And if you don't already have a 12, don't buy one! Buy a ten and build the DR10 instead.

For those of you still awaiting a DR for a 15, it's coming soon. I've built it, it sounds great, but just as the DR10 makes more sense than the DR12a, the DR12 is a better choice than the like-sized DR15a. Of course, that means little if you already have 15″ drivers on hand. If you do, be patient—your true "State-of-the-Art" box is on the way. ❖

Bill Fitzmaurice's earlier speaker designs, including his work with folded horns, are collected in his book *Loudspeakers for Musicians* (available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, 888-924-9465, custserv@audioXpress.com, $9.95).

Updating Borbely's Line Amp

If you find it's time for a new preamp, try this upgrade of a proven performer. **By Glenn Wallberg**

Many years ago I decided it was time for a preamplifier upgrade. The Lampton-Zukauckas Mark I (*Audio Amateur* 1/79) that I constructed back in the early '80s was becoming a bit dated. All the tantalum coupling capacitors, examined with respect to all the audio research of the last 20 years, indicated that an upgrade should lead to better sound. So I developed my list of objectives for a new preamp:

- Line stage only. I haven't used a turntable for quite a while and don't anticipate using it in the future. Not that it doesn't sound good, but I didn't like the bad ergonomics and the fuss of vinyl even when it was the only show in town.
- Straight wire with gain topology. I never used the tone controls on my L-Z preamp anyway.
- Need to fit everything into a Sescom 17″ × 7″ box. I like the cabinets from Sescom for their nice appearance, reasonable prices, and easy machining of aluminum.
- Readily available components from suppliers Mouser or Digi-key. Using standard components provides economy now and ensures that replacement parts—should they be needed—should be available ten or 20 years from now.
- Printed circuit board assembly. I'm not ready to throw away the biggest advance in electronic assembly of the last century.
- A design with some pretensions of quality. My home hi-fi doesn't pretend to be state-of-the-art, but neither is it shabby. My other components are the Borbely 60W amplifier and ADS 710 speakers. Signal source is tuner, CD player, or DVD player.

I looked through back issues of *Audio Electronics* for designs that met my requirements. I have enjoyed my Borbely amplifier for many years and so looked at his preamp designs first. But the difficulty of obtaining FETs turned me away from these designs. Other designs had printed circuit boards too large or parts difficult to obtain.

Having failed to find a suitable existing design, I looked next for a design that I could adapt to fit my requirements. This follows the first rule of professional engineering (professional referring here to on the job rather than in the classroom)—never do anything original unless you have to.

Examining again the design of the original Borbely preamp line amp, from *Audio Amateur* 4/85 and 1/86, I decided that I could use this design by replacing the FET input stage with a bipolar input stage. The constant-current sources feeding the input stage make input transistor biasing non-critical (this is not true with his later designs). Adapting this design to bipolar input stage is only a matter of component value changes to achieve suitable currents through the various stages.

Having decided on a design, I proceeded to build it (*Photo 1*). And then I liked it so much I built a second one just like it (*Photo 2*). And now, finally, I would like to share it with others. I will not spend too many words describing sound quality. It is a clearly audible improvement over the old L-Z Mark I design.

I lent it to my brother for a while and he reported it is clearly superior to his NAD preamp-tuner. An electronic tech friend performed some simple tests on it. He reported that the noise floor was below his test bench and that it beautifully reproduced 10V peak-to-peak

PHOTO 1: Borbely preamp shown with tuner and amplifier. Amplifier is a Borbely 60W, built using a Dynaco 150 chassis. The preamp power transformer is contained in the small enclosure above the amplifier.

square waves at 10kHz. I suspect its performance is probably not far from the Borbely design it is adapted from.

COMPONENT CHANGES

The changes required from the original design involve replacing the input transistors and components that set standing current in each stage (*Fig. 1*). For bipolar input stage a first stage current of 2mA (1mA per transistor) is desired with second stage current of 10mA, compared to 10mA current in both first and second stages used with the FET input stage. Erno Borbely explains how to adjust the currents for his topology in his various amplifier articles ("Servo 100," *Audio Amateur* 1/84 and 2/84). Since the standing currents used here are the same as the Servo 100 design, the following parts substitutions are used based on that design (designations are the same as the line amp design of 1/86):

- Q1 becomes two NPN bipolar transistors and Q2 becomes two PNP bipolar transistors. The two pairs should be complementary and matched, if you have a transistor tester available. I used complementary pairs of type 2N5210 (NPN) and 2N5087 (PNP) in one preamp, and MPS A06 and MPS A56 in the other preamp.
- Despite the lower gain of the MPS Ax6 transistors, the preamp built with these transistors had lower output offset (before installation of the DC servo). This is because of limitations of my transistor tester. My old

PHOTO 2: The second Borbely preamp. Same except for the optional Sescom black anodized face plate.

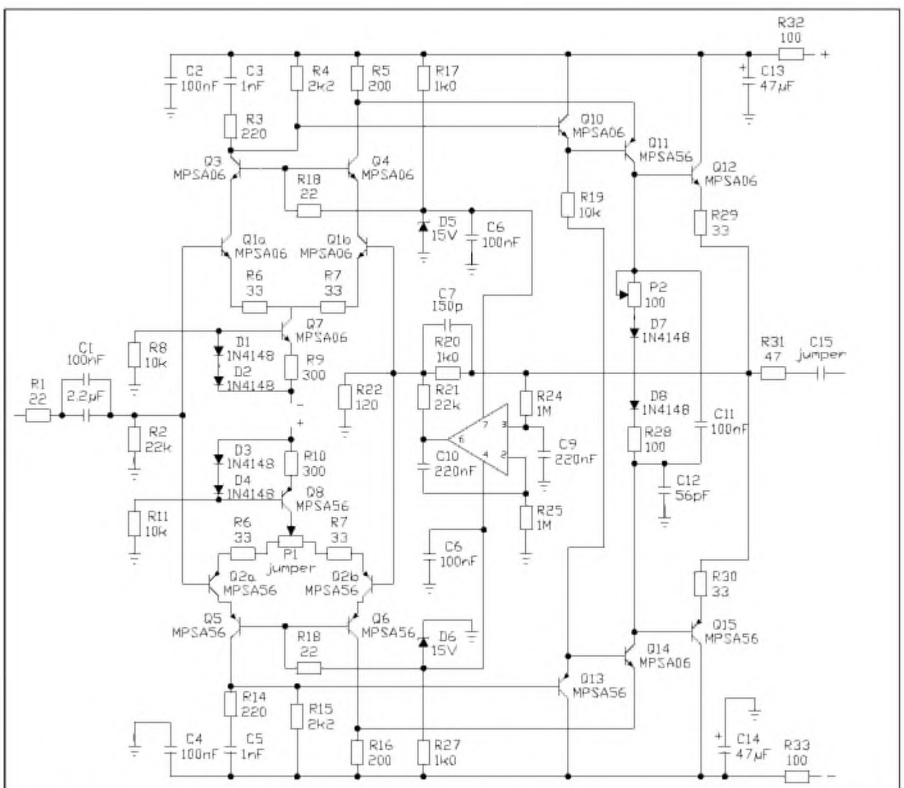

FIGURE 1: Redrawn circuit diagram showing changed component values.

Heathkit transistor tester has trouble accurately measuring transistor gain of high gain components, and so I was able to do a better job matching the MPS Ax6 devices.

- The current for the input stage is set by D2, D4, R9, and R10 (I = Vdiode/R). Using LM336 for D2 and D4, the resistors R9 and R10 become 2k5 to set the first stage current at 2mA (1mA per input transistor). Alternatively, I used a 1N4148 diode (connected in opposite direction) rather than LM336 for D2 and D4; R9 and R10 then become 300.
- R4 and R15 become 2k2 and R5 and R15 become 200 to set the current gain of the second stage at ten.
- R3 and R14 become 220, and C3 and C5 become 1000pF (copied from Servo 100 design).
- R2 becomes 22k to better provide bias current for the input stage.
- I did not use the balance potentiometer R23, but replaced it with a jumper, and R20 becomes 1k0.
- I did not use the output coupling capacitor C15, but, again, replaced it with a jumper.
- I retained the input coupling capacitor C1. The bipolar input stage will inevitably have more input offset voltage than a FET input stage because of the bias currents, and even small voltages can lead to degradation of the volume control potentiometer.
- I did not use offset adjustment potentiometer P1, jumpering it out and using identical values, 33Ω, for R6, R7, R12, and R13. This is at the discretion of the builder.

All other components are according to the original article.

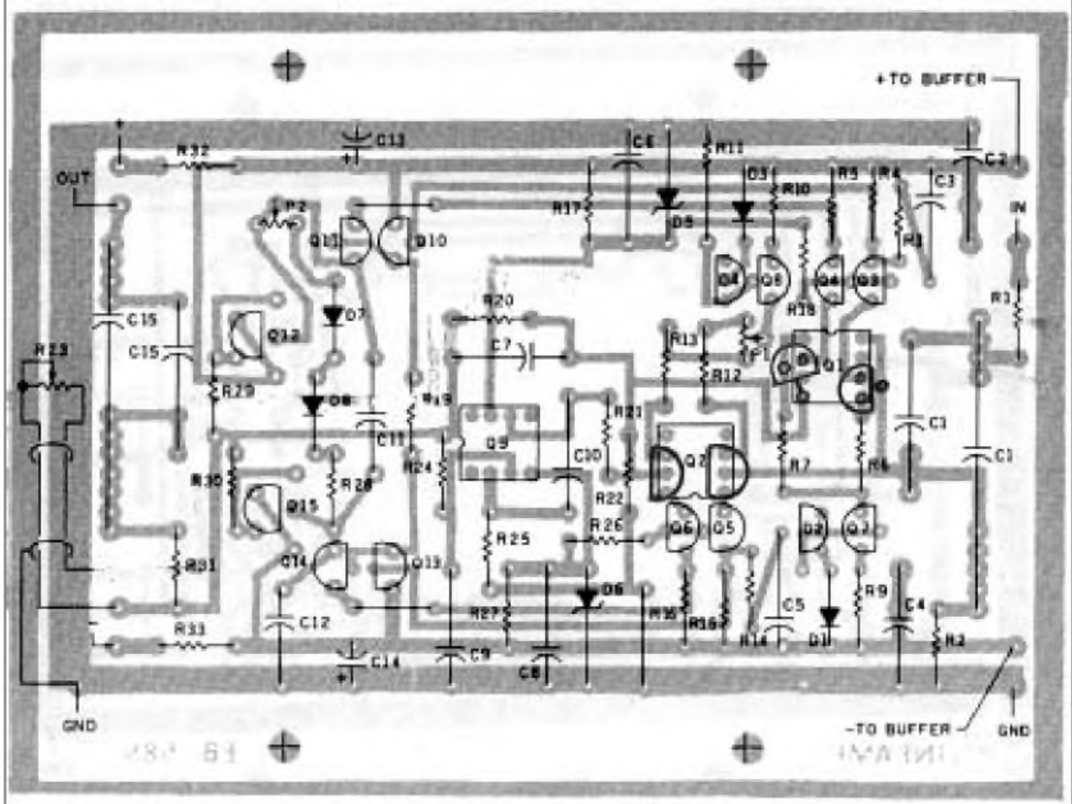

FIGURE 2: Revised stuffing guide.

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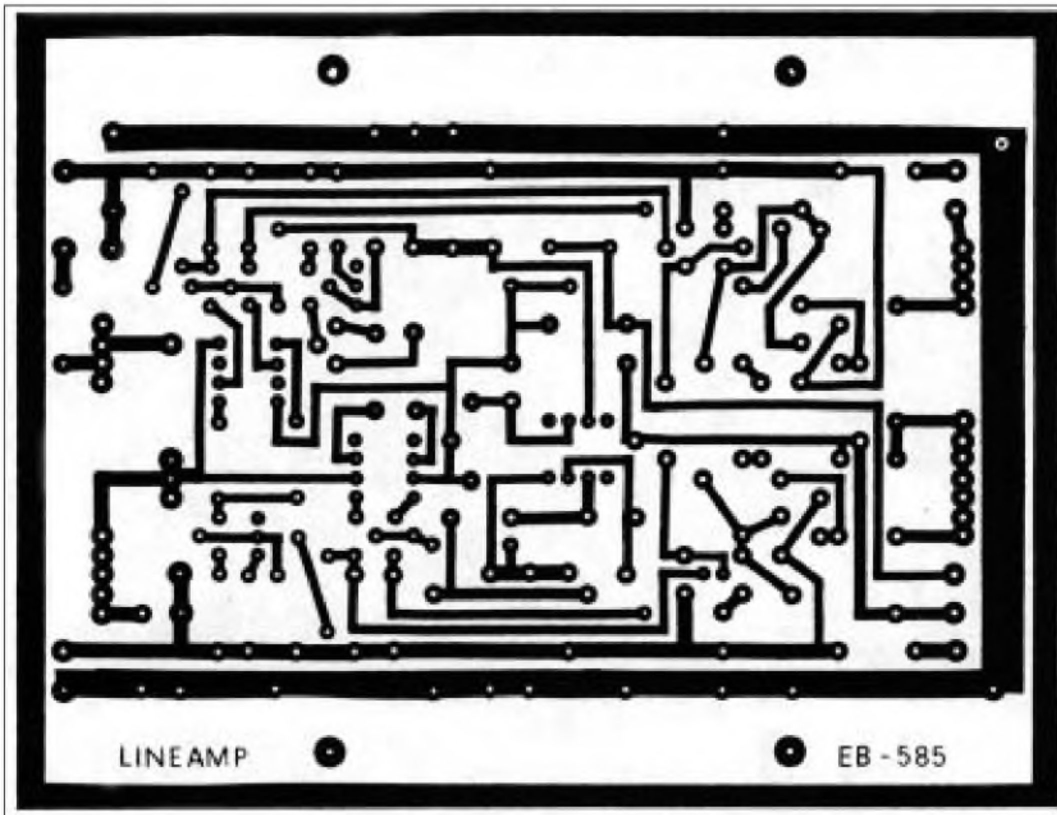

FIGURE 3: Circuit board layout for the line amplifier.

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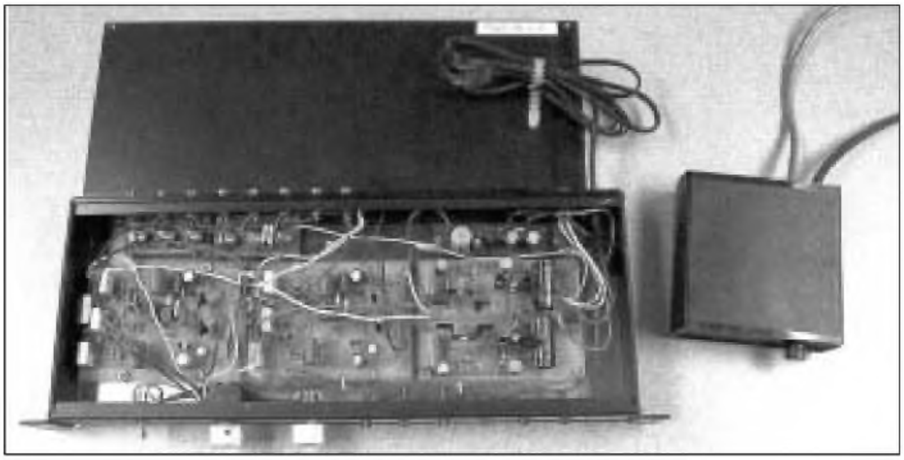

PHOTO 3: Interior of Borbely preamp. The three main circuit boards are the two Borbely line amps and the Borbely improved power supply. The input switching is performed by small Omron relays located adjacent to the input jacks. A small power-supply board located in the back right provides 24V DC used by the input switching relays, the indicator lights, and a power relay located to the right used for the switched outlet.

MAKING THE CHANGES

You can use the original circuit board design. This board is no longer sold by Old Colony—the people at Old Colony informed me that I purchased the very last one in stock several years ago when I built my second project. With the original circuit board design, the following changes are required for mounting the bipolar input stage. [Send an accurate photocopy of Fig. 3 to Atlas Circuits Co., PO Box 892, Lincolnton, NC 28092. Include a color coded second copy indicating hole sizes. The cost is 69¢ per square inch or $16.13, plus 2¢ per hole and a $6 fee for photo art, plus $4 postage. You may pay with a credit card.—Ed.]

- Q2 becomes Q2a and Q2b, two PNP transistors. The standard e-b-c pinout of TO-92 bipolar transistors fit directly into the corresponding s-g-d terminals of the AH5020CJ pinout.
- Q1 becomes Q1a and Q1b, two NPN transistors. Unfortunately, the pinout of the bipolar transistors does not correspond to the pinout of the NPD5566 device. By drilling additional holes in the circuit board for transistor leads, it is possible to mount the transistors without mutilating their legs.

See revised stuffing guide (*Fig. 2*) for transistor mounting. *Figure 3* shows the circuit board layout.

POWER SUPPLY ISSUES

This is not an issue—the Borbely improved power-supply design (based on the Sulzer design, "A Moving Coil Preamp, Pt. 2," *Audio Amateur* 1/87) has my highest recommendation. This project was the first time I used this regulator, and this preamp is the first I have had that is absolutely quiet. With the signal source off, it is impossible to discern whether the hi-fi is turned on or off, even with your ears directly next to the cones of the speakers.

I used a single power supply for both channels. In my implementation of this supply I could not locate a source for transistor types BD241A and BD242A and replaced them with TIP29C and TIP30C. The circuit board is also available from Old Colony.

The switched outlet and input selection use relays with a separate power supply. I used a three-terminal power regulator for this application, which is probably overkill. The relays coils are inductive loads, so the regulator is protected by reversed diodes. Using the power relay for the switched outlet lets me use a mini-toggle switch as the front panel on-off switch rather than a larger switch adequate for the power amp.

The preamp selector switch is a standard 2P6T rotary switch. One of the poles is used to select the input relay; the other pole is used for the front panel indicator light selection. The volume potentiometer is an Alps Blue supplied by Old Colony (PO Box 876, Peterborough, NH 03458, 603-924-9465, Fax 603-924-9467, custserv@audioXpress.com; limited supply available). It is a pleasure to use such a nice quality control. The preamp power transformer is external, connected by an umbilical cord.

SUGGESTIONS

I recommend the Sescom enclosures, but they do not provide sufficient shielding without modification. The panels are held in place by the rails and the anodizing of the rails isolates the panels. This is not satisfactory for a preamp that requires shielding to avoid noise pickup. I made sure that all panels had a lug with wire connection to other panels. Another practical suggestion for builders of preamps is to be sure to supply front-panel indicator lights from a DC source.

FUTURE IMPROVEMENTS

One change I would recommend if you repeat this design is to redesign the circuit board layout so you can install the bipolar transistors Q1a and Q1b straight and true. For the ultimate development of the design, see borbelyaudio.com. ❖

The Extreme Subwoofer

Besides making one young audiophile happy, this speaker project explores a cabinet bracing strategy and a low Q sealed box approach. **By Jim Moriyasu**

This subwoofer project was a birthday gift for my 14-year-old son, Thomas, who enjoys skateboarding. He dubbed it "the Extreme subwoofer" because of its size and capability. I also kid him about it as a young audiophile's rite of passage. After all, how many teenagers can play Eminem at 90dB with low-end response to 25Hz?

GETTING IT TOGETHER

Thomas currently has a pair of mid-sized monitors that were recycled from an old pair of small Advent enclosures. They now house an Audio Concepts AC-8 8″ woofer paired with an MB MCD25AV 1″ titanium dome tweeter. Since the enclosures are on the small side, the AC-8 woofers have an f_3 of about 80Hz, while the box Q comes in around 0.9. It's a very nice-sounding loudspeaker despite the diffraction from the grille, but, of course, it lacks the last octave of bass.

So a subwoofer with a built-in amplifier would be a great help. I chose the Keiga KG-5150-V from Meniscus, which features 150W, a low-pass crossover variable between 50Hz and 100Hz, and a bass boost of 4dB at 25Hz. Meniscus sells it for $169.

The amplifier also has speaker level and low-level inputs. The low-level inputs allow connection to an A/V receiver's subwoofer out jack(s), while the high-level inputs allow direct connection from a receiver or power amplifier's speaker output jacks. With Thomas's speakers, we connected the left and right loudspeaker outputs from his receiver to the subwoofer amplifier, then connected his speakers to the subwoofer amplifier's high-level outputs.

I find these powered subs to be a very convenient and effective way to add a subwoofer to a system with small loudspeakers. The only drawback is that there is no high-pass filtering to the satellites, although you could construct a passive high-pass crossover, if power handling were an issue.

Meniscus also supplied the SV-12 woofer, one of the best 12″ drivers in its class. It is the improved version of the DV-12, which has been used in several *Speaker Builder* articles. The driver features a rigid long-fiber ribbed cone that has a resonant frequency of 17.5Hz and linear excursion of 13mm.

With two stacked magnets weighing more than 4 lb and a 2″ 350W voice coil, Q_{TS} comes in about .40, so it works well in sealed boxes from 2 to 6ft^3. It's all held together by a very rugged-looking aluminum cast frame and it comes with a mounting gasket and heavy-duty gold-plated solderless terminals. It is well worth the $150 price tag.

EARLY TESTS

Earlier studies of panel damping showed bracing to be a particularly effective way of reducing panel resonances by 5–10dB. To further test the effectiveness of bracing, I built the Extreme subwoofer cabinet with two kinds of bracing: shelf bracing with a single, large hole and vertical rails that are ½″ thick and 2″ wide. I built a test cabinet with exactly the same dimensions without bracing and used it for comparison tests.

PHOTO 1: Side and rear view of the Extreme subwoofer showing the input panel and heatsink of the Keiga KG-5150-V power amplifier.

I ran simulations on Loudspeaker Enclosure Analysis Program (LEAP), by LinearX, with box volumes of 2, 3, and 4ft^3. Since I wasn't concerned about compactness and wanted to try a sealed box with a Q of less than 0.7, I decided to go with the largest volume. This approach resulted in an enclosure with internal dimensions of 15″ wide × 15″ wide × 30.75″ tall.

I decided to mount the woofer facing downwards and 3″ from the floor. Earlier tests showed this configuration would not have any detrimental effect

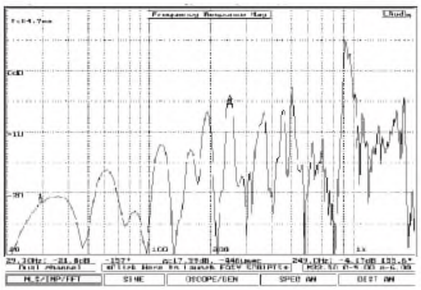

FIGURE 1: Output from the ACH accelerometer showing resonance peaks from the side panel of the unbraced test cabinet.

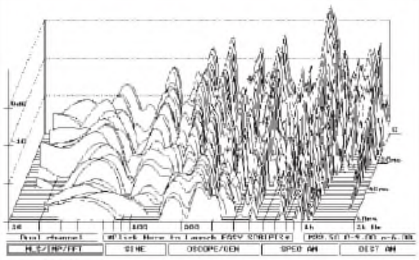

FIGURE 2: Cumulative spectral decay version of the output from the ACH accelerometer attached to the side panel of the unbraced test cabinet.

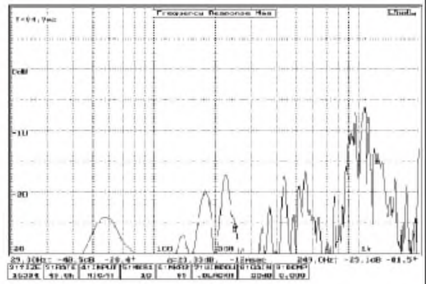

FIGURE 3: Output from the ACH accelerometer showing resonance peaks from the side panel of the Extreme subwoofer enclosure.

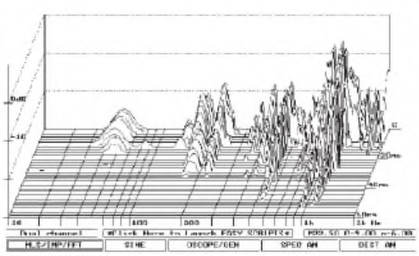

FIGURE 4: Cumulative spectral decay version of the output from the ACH accelerometer attached to the side panel of the Extreme subwoofer enclosure.

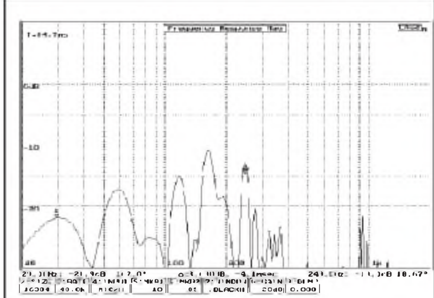

FIGURE 5: Output from the ACH accelerometer showing resonance peaks from the side panel of the unbraced test cabinet with a 100Hz low-pass crossover.

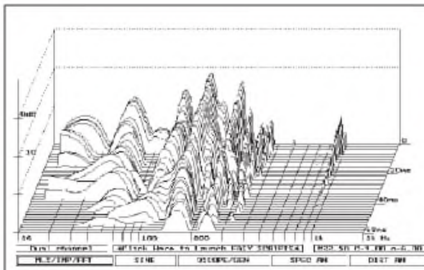

FIGURE 6: Cumulative spectral decay version of the output from the ACH accelerometer attached to the side panel of the unbraced test cabinet with a 100Hz low-pass crossover.

on the woofer's output. I also thought it would be aesthetically interesting to have a black monolith rather than a fabric-fronted box. I stuffed the enclosure with 4 lb of Acousta-stuf, a crimped fiber Dacron material.

I also installed an internal sub-enclosure for the amplifier. While you can install the amplifier without a sub-enclosure, there is still the possibility of air leaks from the seals around the switches and potentiometers. So a sub-enclosure is the best bet for an airtight enclosure. The sub-enclosure requires internal dimensions of 7¼″ wide × 10½″ high × 3¾″ deep.

ASSEMBLY

Photo 1 shows the Extreme subwoofer. We sprayed it with black epoxy enamel. Thomas likes to do graphic design, so eventually it will become more than a black monolith. We used 1″ by ⅛″ extruded aluminum right angle for the side rails and legs. The aluminum is commonly available at hardware stores.

Photo 2 shows the half-inch, 9-ply, Baltic birch plywood bracing along

PHOTO 2: An upside down and inside view of the Extreme subwoofer enclosure showing the four shelf braces, four vertical rails, and the amplifier sub-enclosure.

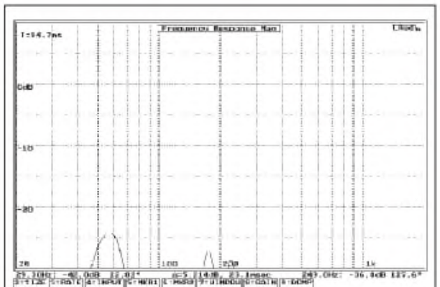

FIGURE 7: Output from the ACH accelerometer showing resonance peaks from the side panel of the Extreme subwoofer enclosure with a 100Hz low-pass crossover.

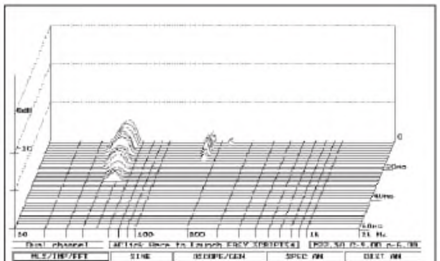

FIGURE 8: Cumulative spectral decay version of the output from the ACH accelerometer attached to the side panel of the Extreme subwoofer enclosure with a 100Hz low-pass crossover.

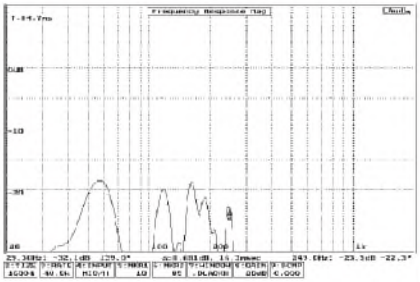

FIGURE 9: Output from the ACH accelerometer showing resonance peaks from the side panel of the unbraced test cabinet with a 75Hz low-pass crossover.

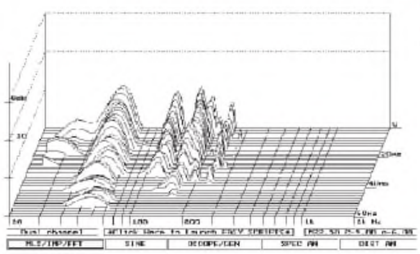

FIGURE 10: Cumulative spectral decay version of the output from the ACH accelerometer attached to the side panel of the unbraced test cabinet with a 75Hz low-pass crossover.

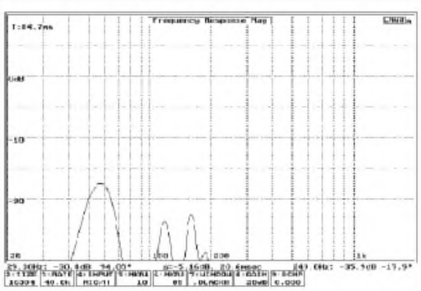

FIGURE 11: Output from the ACH accelerometer showing resonance peaks from the side panel of the unbraced test cabinet with a 50Hz low-pass crossover.

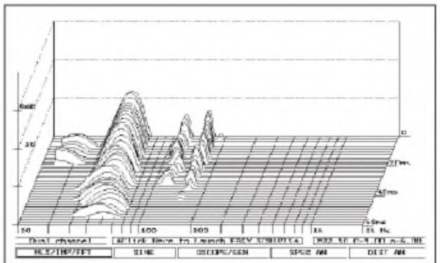

FIGURE 12: Cumulative spectral decay version of the output from the ACH accelerometer attached to the side panel of the unbraced test cabinet with a 50Hz low-pass crossover.

with the enclosure for the power amplifier. There are four shelf braces, 15 in^2 with various cutouts. The basic cutout is an 11″ diameter circle. However, the brace across the amplifier enclosure forms a U while the two braces that form the sides of the amplifier enclosure form a D. The vertical rails and the shelf braces are joined with a cross-lap joint, which we produced using a dado-ing blade on a table saw. We glued the bracing structure together first, then assembled the enclosure around it. We used yellow carpenter's glue and biscuits for ease of assembly and strength.

MEASUREMENTS

Let's start with the test cabinet. I began this project with the expectation that the test cabinet would have severe resonances, while the Extreme cabinet, with its multiple braces, would have similar modes, but they would be reduced in magnitude by 5–10dB. And, I knew that most of the vibrational modes in MDF panels occurred above 100Hz, so I hypothesized that the additional bracing wouldn't be necessary because the subwoofer output is rolled off above 100Hz.

Effective design and engineering also means using less material when using more would be superfluous. While cabinets that do well on the knuckle-rap test because they have extra-thick walls and lots of bracing are great for marketing, maybe we could save ourselves from extra woodworking, higher material costs, and strained backs by making simpler subwoofer cabinets.

Figure 1 is from the output of an ACH accelerometer taped to the middle of the test cabinet's side panel with the low-pass crossover switched off. I measured the output with Liberty Instruments' Liberty Audio Suite measurement system. Obviously, there are lots of resonance peaks, which become more obvious in *Fig. 2*, which is the cumulative spectral delay (CSD) version of *Fig. 1*. Major resonances occur at 60Hz, 120Hz, 195Hz, 250Hz, 370Hz, 500Hz, and 900Hz. The resonances take up to 60ms to decay 20–30dB.

The effectiveness of the bracing in the Extreme cabinet is obvious by examining *Figs. 3* and *4*. Resonances are reduced in number while the levels are reduced by 5–10dB.

With the crossover switched in and set to 100Hz, however, you see in *Fig. 5* that resonances above 300Hz are significantly reduced. There are still troublesome resonances at 120Hz, 170Hz, and 250Hz, however, and these are still 5–10dB higher than in the Extreme cabinet.

Figure 6 is the CSD view that shows how persistent the resonances are. With the 100Hz crossover, the Extreme cabinet shows just an insignificant peak at 170Hz (*Figs. 7* and *8*). *Figures 9* and *10* show the test cabinet results with a 75Hz low-pass crossover. The vibrations above 100Hz are now reduced by 5–10dB; however, the 60Hz mode is down only a couple of dB.

Figures 11 and *12* show the test cabinet with a 50Hz low-pass crossover; here, resonances are reduced another 3–4dB and probably could be considered harmless. These results suggest an unbraced subwoofer cabinet might be adequate if you use a low-pass crossover below 75Hz.

Are these resonances high enough in level to be heard? To find out, I conducted ground-plane sound pressure level (SPL) measurements at 1m with 4W. *Figure 13* compares the test cabinet with the Extreme cabinet without a low-pass crossover. The test cabinet has 0.20ft^3 of additional volume, since it doesn't have an enclosure for the KG-5150-V amp, so its output below 100Hz is a little over 1dB better than the Extreme box. (For the record, the test box with the SV-12 was down 3dB at 24Hz, the box resonated at 28Hz and had a Q_{TS} of 0.64, while the Extreme/SV-12 combination was 27Hz, 27Hz, and 0.57, respectively. I measured these with my NAD2040 power amp and not the KG-5150-V amp.)

The difference between the two boxes was raised by 65dB. The bumps and dips of the difference curve don't correlate exactly with the resonance measurements; however, it would be safe to say that the box resonances are affecting the output above 100Hz. The 60Hz resonance does not appear to be high enough in level to affect the out-

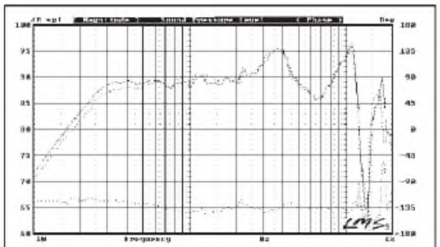

FIGURE 13: Ground-plane sound pressure level comparison of Extreme subwoofer versus unbraced test box run full range. The difference between the two curves is raised by 65dB and is the solid line above and below 65dB.

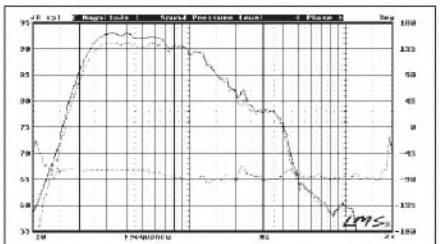

FIGURE 14: Ground-plane sound pressure level comparison of Extreme subwoofer versus unbraced test box with KG-5150-V amp and 100Hz low-pass crossover. The difference between the two curves is raised by 65dB and is the dotted/dash line above and below 65dB.

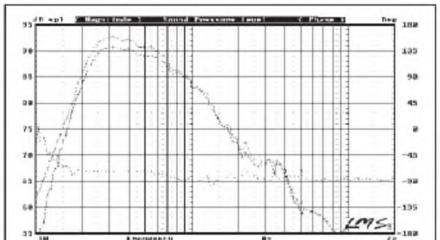

FIGURE 15: Ground-plane sound pressure level comparison of Extreme subwoofer versus unbraced test box with KG-5150-V amp and 75Hz low-pass crossover. The difference between the two curves is raised by 65dB and is the dotted/dash line above and below 65dB.

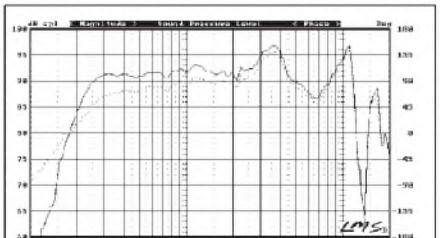

FIGURE 16: Ground-plane sound pressure level comparison of Extreme subwoofer with KG-5150-V power amplifier without a low-pass crossover (solid line) versus the NAD2140 power amp.

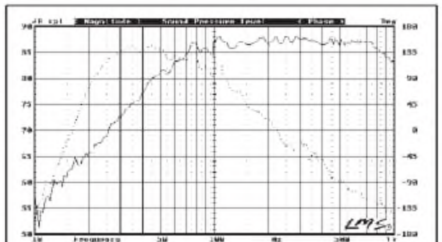

FIGURE 17: Individual ground-plane sound pressure level responses of Extreme subwoofer with low-pass crossover (dashed line) and AC8/MB satellite (solid line).

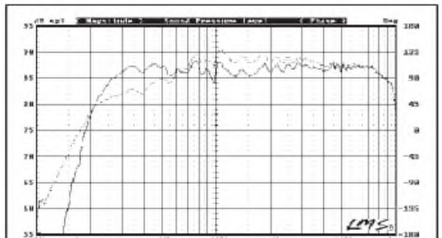

FIGURE 18: Summed ground-plane sound pressure level responses of Extreme subwoofer with low-pass crossover and AC8/MB satellite. The in-phase combined response is the solid line, while the reverse phase is the dot/dash line.

put below 100Hz.

I then switched amps to the KG-5150-V and switched on the low-pass crossover. With the 100Hz crossover (*Fig. 14*), output is down 5dB at 150Hz and 8dB at 200Hz. The difference curve, raised 65dB, shows the effect of the resonance on the test box's results. It appears that the 120Hz resonance could still be a problem.

Figure 15 shows the effect of the 75Hz crossover. Output is down by 6dB at 100Hz and 18dB at 200Hz. Resonances generated by the test box are mostly out of the picture with this level of filtering. These results corroborate the results shown in *Figs. 11* and *12*.

PERFORMANCE

For those interested in how well this subwoofer performs, I've included the information in *Figs. 16–18*. *Figure 16* shows the output of the KG-5150-V without the low-pass filter versus the NAD test amplifier. The KG-5150-V provides 4dB of boost at 25Hz. Initially, I thought it only affected frequencies below 100 cycles. However, this chart shows that its effect goes out to 1kHz!

Figure 17 shows the individual outputs from the AC8/MB satellite and the Extreme subwoofer with the low-pass crossover switched on. Their responses cross over at 75Hz. It took a couple of tries to get to this result by fiddling with the subwoofer output and crossover frequency.

Figure 18 shows the summed output of the satellite and the subwoofer with normal phase and reverse phase. With normal phase the –3dB point is 23Hz and the response is almost flat through the crossover region. Oddly enough, below 20Hz, output is reduced due to interaction with the satellite's woofer. Still, the crossover isn't optimal because the reversed phase condition produces up to a 4dB dip between 20–60Hz, as much as 2dB of boost above 60Hz and below 20Hz the system's output rises due to interaction with the satellite's woofer output.

CONCLUSION

Thomas really enjoys his sound system, as you would guess. Is it just a "guy" thing that we love deep bass, that we aren't as sensitive as the fairer gender, and must feel the floor shake to really enjoy music? The engineer in me says we must reproduce music as it was recorded. The guy in me says, "more power, bigger speakers, deeper bass, argh, argh, argh..."

But seriously, getting that last octave of bass is worth it, and with subwoofer amps and subwoofers like the KG-5150-V and SV-12, it is much easier and affordable to do than ever before. And if you are using satellites that go down to 80Hz, you should be able to build and use a simple, unbraced cabinet that is cheaper and easier to build and to move around. ❖

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moriyasuj001@hawaii.rr.com

SOURCES

Meniscus Audio Group, Inc.
(616) 534-9121
www.meniscusaudio.com

LinearX
(503) 612-9565
www.linearx.com

Liberty Instruments
(513) 755-0252
www.libinst.com

Octal to 7-Pin Tube Adapter

When you need an adapter to test 7-pin tubes, make your own.

By Charles Hansen

PHOTO 1: Octal to 7-pin adapter.

I needed to test a few 7-pin tubes, and I didn't have a conventional tube tester. The Audiomatica Sofia tube curve tracer I use does not have a 7-pin miniature tube socket. This is understandable, given the large number of 7-pin tubes and their nonstandard pin assignments. My solution was to make an octal to 7-pin adapter (*Photo 1*). I understand these general-purpose adapters were readily available at one time, but I could not locate any current source. (Specific adapters are still available: EL84 to EL34 sockets, 6GM5 to 7591 sockets, and so forth.)

The exploded view of my adapter is shown in *Fig. 1*, along with the Antique Electronic Supply (480-820-5411, www.tubesandmore.com) part numbers I used. Parts cost $4.

HOW IT WORKS

The Sofia uses jumper cables to simultaneously connect its test signals to all same-numbered pins of its four test sockets. The Sofia has a limited number of tube models–all audio tubes–to choose from. When you select a tube model from the Test Menu, the test model software determines the filament voltage, grid voltages, and plate voltage and current. With the appropriate adapter, you could plug a tube into any socket on the Sofia (assuming enough pins are available).

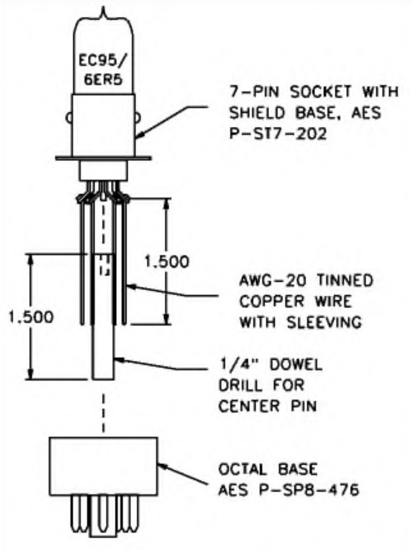

FIGURE 1: Octal to 7-pin adapter exploded assembly view. G-2131-1

Therefore, I wired the seven pins on the miniature socket to the corresponding pins on the octal base. I brought a free lead from octal socket pin 8 to the side of the octal base.

CONSTRUCTION

I chose a 7-pin ceramic socket with a shield base. This gave me a means to grip the socket to install and remove tubes without placing any strain on the wiring between the 7-pin socket and octal base.

The octal base pins accept 20-gauge

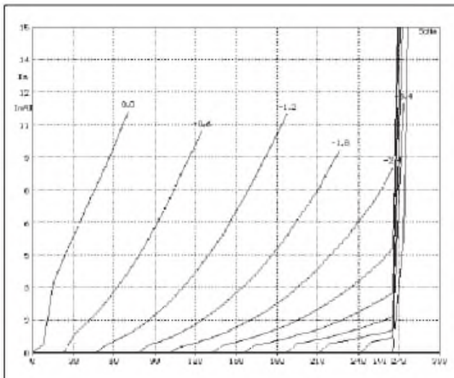

FIGURE 2: Sofia curves for 6ER5, using 6SL7 model with shunt zeners. G-2131-2

solid wire, so I soldered a 2″ length of #20 tinned bus wire onto each lead of the 7-pin socket. Then I slipped a ¾″ long piece of insulating sleeving over each wire for safety. The 7-pin socket has a center pin, which you can use as a free wiring terminal point.

I center-drilled a ¼″ wooden dowel, 1.5″ long, with a 5/64 drill bit. The hole in the dowel accepts the 7-pin socket center pin. The other end of the dowel fits into the hollow locator pin of the octal socket. The dowel serves as a support strut for the whole assembly.

Before assembling the socket and base, I soldered a 2″ length of #20 wire into pin 8 of the octal base and routed it over the outside. Next, I placed the 7-pin socket on the dowel, and inserted the free ends of the seven wires through the corresponding hollow pins of the octal base. Then I pushed the lower end of the dowel into the octal socket locator pin.

Finally, I cut the seven wire leads poking through the octal base pins flush, and soldered them to the pins. In order to prevent solder from coating the outside of the pins, I pushed the solder from the inside of the socket while heating the bottom of the pins. In this way, only a minimum amount of solder flows outside the octal base pins.

USING THE ADAPTER

I wanted to test 6ER5/EC95 tubes, which have a maximum Vp of 250V. Since there is no Sofia tube model that limits Vp to so low a voltage, I decided that for the safety of the tubes, I should find out how the Sofia delivers its plate voltage. I installed a 12AX7, which has a Vp maximum of 330V, and ran a standard plate curve test.

My next test involved installing a 1N4981 91V 5W zener in series with the plate test jumper. When I ran the 12AX7 test program, I found the curves went out to almost 400V at the most negative grid voltages. I suspect the Sofia monitors the actual applied plate voltage, and increased its supply voltage to overcome my series zener voltage.

In my final test, I connected three 1N4981 zeners in parallel with the plate-cathode leads of the 12AX7 using a kludge of clip leads and the two Sofia jumpers used for screen and suppressor grids. This parallel zener string limits the voltage that the Sofia can apply to the plate of the 12AX7 to a nominal 273V. I didn't see how this could harm the Sofia, because it is designed to withstand a shorted tube.

Now the plate voltage was clipped at about 265V on the right side of the plate curves, but the data below clipping matched the original data on the unmodified 12AX7 test setup. The zener leakage is only 2µA, so the shunt zeners don't affect the plate curves.

Since I intended to run the 6ER5s at fairly low current, I selected the 6SL7 model (Vp maximum is 300V), and connected the three 1N4981 zeners from pin 5 (Pt) to pin 7 (K) of the 7-pin socket with external clip leads. The plate curves clipped at a safe voltage for the 6ER5 (*Fig. 2*), and my 12AX7 tests suggest that this doesn't limit the ability to read all the tube parameters at selected operating points below clipping.

Other 7-pin tubes, such as the 12AV6 with plate curves identical to the 12AX7 model, can be tested without a parallel zener string. You can also use pin 8 of the octal socket to easily install components in series with any of the other 7-pin socket connections. ❖

Product Review
The KAB Preamp

Reviewed by Gary Galo

K-A-B Souvenir EQS MK12 Disc Remastering Preamplifier. K-A-B Electro-Acoustics, PO Box 2922, Plainfield, NJ 07062. 908-754-1479, FAX 908-222-3442, info@kabusa.com, www.kabusa.com. EQS MK12–$945. Warranty: One year.

PHOTO 1: The K-A-B EQS MK12. Twelve equalization curves accommodate records from early acoustics through RIAA stereophonic discs. Adjustable cartridge loading, lateral-vertical switching, and a processor loop add to the unit's flexibility.

PHOTO 2: Rear panel of the EQS MK12. The ¼″ jacks on the left are fully-balanced, tip-ring-sleeve outputs. RCA connectors are gold-plated and Teflon®-insulated. The jacks marked "Return 1" are actually the line inputs, which are placed ahead of the equalization circuits.

K-A-B are the initials of Kevin A. Barrett, proprietor of K-A-B Electro-Acoustics, a small audio company based in New Jersey specializing in analog disc playback. K-A-B carries a full line of products for playing disc records of practically every type, including turntables, custom stylii, signal-processing devices, preamplifiers, and a wide variety of accessories. Their website is a "must visit" for anyone who collects 78s and other types of historical recordings. The EQS MK12, K-A-B's top-of-the-line stereo phono preamplifier, has been designed to provide user-selectable playback equalization for records pre-dating the RIAA standard, including 78-rpm discs, early LPs, and program transcriptions (*Photos 1* and *2*).

FEATURES

The EQS MK12 has two phono inputs, selectable at the front panel. Many collectors will have separate turntables dedicated to 78-rpm records and LPs–the EQS MK12 allows both to be connected. You can adjust cartridge loading at the front panel with trimmer capacitors that are continuously variable between 20pF and 200pF. A 12-position rotary switch selects the resistive loading, adjustable between 100Ω and 100k.

Twelve equalization curves are supplied to accommodate the myriad recording curves used throughout the history of the disc record. K-A-B calls this a "Chronologic Equalizer" since the push buttons are arranged as a record equalization timeline, advancing through the history of equalization curves from left to right. The transition points for the curves are shown in *Table 1*.

Barrett has based his selection of the various turnover points on practical experience, as well as data supplied in the *Radiotron Designer's Handbook* (CD-ROM and printed versions are available from Old Colony; Chapter 17 is required reading for those interested in these matters). Stanley Lipshitz's landmark Audio Engineering Society paper was used to calculate the filter designs (another required reading).[1] A discussion of the nature of disc recording equalization is beyond the scope of this review. For an overview, I suggest my paper "Disc Recording Equalization Demystified," reprinted in *The LP is Back!*.[2]

In *Table 1*, f_3, f_4, and f_5 refer to the three transition points, as labeled by Lipshitz. The low-bass turnover is f_3, the bass turnover is f_4, and the treble transition frequency is f_5. The AC curve is for acoustic records, and AE is for very early electrical discs (such as Victor electrics from 1925 that still bear the wing-style acoustic labels). Curves E3, E5, and E7 are for the bulk of electrically-recorded 78-rpm discs made from the mid-1920s through the late 1940s.

For these three curves, K-A-B uses their unique "Fine Slope" high-frequency attenuation, rolling off at 3dB-per-octave above a corner frequency of 2120Hz, and shelving at –10dB. Some collectors and transfer engineers prefer to think in terms of the attenuation at 10kHz, rather than the actual treble transition frequency. *Table 2* is a conversion chart which should prove helpful.

A special CO curve complements the late Columbia 78-rpm record. The NAB curve (for National Association of Broadcasters) is the standard for 16″ lacquer transcription discs (often incorrectly called "acetates") used in the broadcast industry. Four 33 1/3-rpm curves cover the original Columbia LP, AES (Audio Engineering Society), the early Decca/London FFRR ("Full Frequency Range Recording"), and RIAA (the Recording Industry Association of America).[3]

The twelfth and last curve on the EQS MK12 is *flat*, which means that it's not really a curve at all. The flat position is useful for a couple of reasons. Acoustical recordings approximate a constant-velocity characteristic throughout their limited frequency range. A constant-velocity recording will yield a flat frequency response when played with a magnetic cartridge (for further details, see my previously mentioned paper).

Therefore, those inclined toward a strictly scientific approach to playback equalization often prefer a flat (i.e., non-equalized) playback. Many collectors, myself included, prefer adding a bit of upper-bass/lower-midrange warmth to acoustic records. If the *flat* position is used, the warmth can be added with an equalizer, preferably a parametric.

EQUALIZATION

Transfer engineers who use digital processing (such as CEDAR) for removing noise on 78-rpm recordings sometimes find that a flat initial playback allows the digital noise removal to work better, before the recording curve is applied. The *flat* position can be extremely useful here. The disc is played without equalization, then fed to the digital processor.

The EQS MK12 also has a line input, just ahead of the equalization circuitry, which allows you to apply the recording curve at line level, after the digital processing has been done. Some engineers will tell you that you can apply the playback curve in the digital domain, and

TABLE 1
MANUFACTURER'S SPECIFICATIONS

Input Section:
Input capacitance: 20–200pF continuously variable
Input resistance: 100Ω–100k in 12 steps
Input sensitivity: 11mV @ 1kHz, RIAA, for 1V output
Input overload: 66mV @ 1kHz, RIAA, for 5.9V output
Fixed front-end gain: 40dB @ 1kHz, RIAA, gain set to "0"

Chronologic Equalizer:

Curve	f_3	f_4	f_5	LF Gain Stop
AC	50	500	5000	Yes (+10dB)†
AE	20	200	2120	No
E3	30	300	2120*	No
E5	50	500	2120*	No
E7	70	700	2120*	No
CO	30	300	1590	No
NAB	40	400	1590	Yes (+17dB)†
LP	50	500	1590	Yes (+13.5dB)†
AES	40	400	2500	No
FFRR	30	300	2120	Yes (+17.5dB)†
RIAA	50	500	2122	No
FLAT	0	0	0	No (Gain fixed)

*K-A-B "fine slope" high-frequency rolloff, 3dB/octave shelving at –10dB.
†Low-frequency gain stops limit the total bass boost to the figure stated; ref. 0dB @ 1kHz.

Rumble Filter:
Corner frequency: 30Hz
Attenuation: 24dB/octave

Output Stage:
Active balanced tip-ring-sleeve: 12V RMS maximum
RCA single-ended: 6V RMS maximum

Distortion and Noise (Ref. 1V out):
THD: <0.05%
IMD: <0.05%
S/N: >–75dB

Physical Specifications:
Dimensions (W × H × D): 19″ × 1.75″ × 8.25″
Weight: 5 lbs
Shipping weight: 8 lbs

TABLE 2
TREBLE ROLL-OFF CHART

10kHz Attenuation	Transition Frequency (and Time Constant)
–5dB	6800Hz (23.41µs)
–8.5dB	4056Hz (39.24µs)
–10dB	3333Hz (47.75µs)
–10.5	3128Hz (50.88µs)
–12dB (AES)	2595Hz (61.33µs)
–13.73dB (RIAA)	2122Hz (75µs)
–14dB	2036Hz (78.17µs)
–15dB	1807Hz (88.08µs)
–16dB (NAB and Columbia LP)	1591.55Hz (100.0µs)
–20dB	1005Hz (158.36µs)

*The formula for converting –dB at 10kHz to the –3dB frequency was generously provided by *aX* regular contributor G.R. Koonce.

the equalization capabilities of many computer-based digital processors and editors will allow you to do this.

There's a flaw in this approach, however. Among the virtues of digital equalization is the lack of nasty phase shifts inherent in analog filters. In the case of playback curves, the lack of frequency-dependent phase shifts presents a problem.

Most readers know that the playback curves provide proper frequency equalization for the record. What is generally ignored—probably because it happens automatically—is the fact that the playback curves also provide correct phase equalization. The filters used to produce the recording curves in the first place caused phase shifts.

When complementary analog playback equalization takes place, the phase response of the record is also corrected. Ideally, a properly equalized record will have flat frequency and phase response (the fact that we can get a credible square wave off an LP test record is proof that we can come pretty close when everything is working properly). To achieve this, old-fashioned analog playback equalization is the only way to go; digital filtering won't accomplish this task. It should certainly be possible to design digital filters to mimic both the phase and frequency characteristics of analog filters, but no one has yet done this commercially for playback equalization of disc records, as far as I know.

The line input on the EQS MK12 is applied directly to the passive equalization circuits, bypassing the first gain block. The equalization circuits apply a low impedance load—around 2000Ω—to the source, which should not be a problem for most professional gear with very low-Z outputs. If it is a problem, K-A-B offers a version of the preamp with a high-impedance line input board—this model is the EQS MK12L, which sells for $1045.

After the Chronologic Equalizer comes the processor section, which has a fourth-order (24dB/octave) rumble filter with a –3dB point of 30Hz, plus a processor loop for connection of an equalizer or other signal-processing device. The stereo/mono switch is used in conjunction with the lateral/vertical switch and the mono mix controls. Most monaural recordings are laterally cut. Vertically-cut recordings include all cylinders, Edison Diamond Discs, most Pathé discs, and some others.

When a stereo cartridge is used to play a laterally-cut monaural record, the left and right channels are summed to mono "in phase." The vertical component of a monaural recording, which is almost entirely noise, is cancelled. When the same cartridge is used to play a vertically-cut recording, the polarity of one channel must be reversed before the two channels are summed to mono. In this case, it is the lateral component (again, mostly

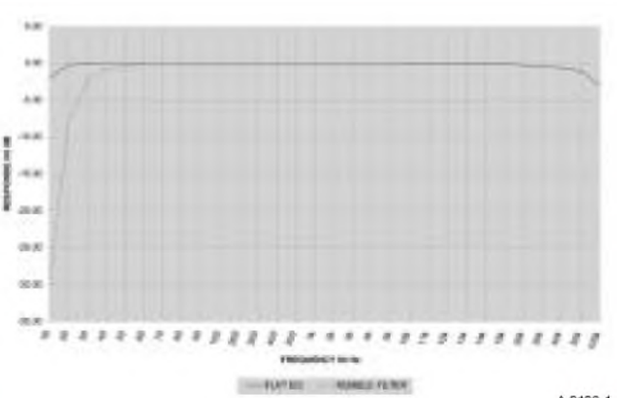

FIGURE 1: Frequency response of the *flat* position, and the rumble filter. The –3dB points of the wideband circuitry extend from below 10Hz to 100kHz.

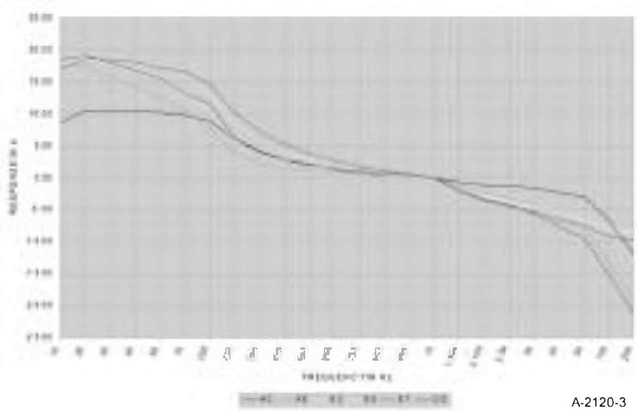

FIGURE 3: Measured response of the six 78-rpm curves. The curves have been intelligently chosen, and will provide musically satisfying listening for most pre-LP recordings.

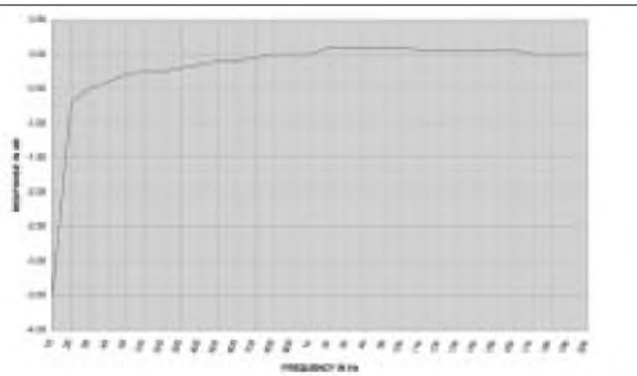

FIGURE 2: RIAA accuracy. The infrasonic rolloff pulls the bass region down slightly, but the response is within ±0.1dB from 500Hz to 20kHz, and ±0.25dB down to 40Hz.

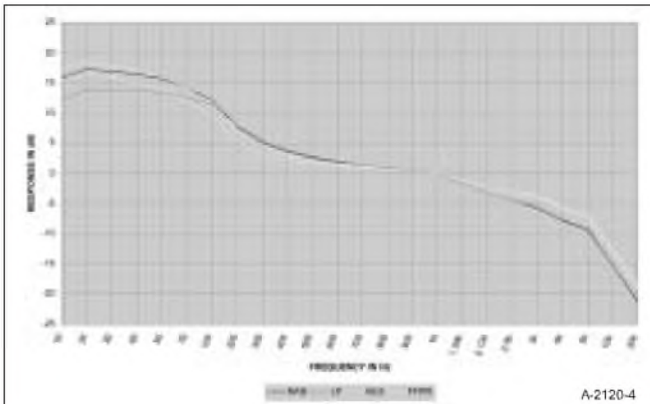

FIGURE 4: Measured response of the four early 33 1/3-rpm curves. These curves accommodate 16" transcriptions, original Columbia LPs, records made with the AES curve, and early Decca/London FFRR discs.

noise) that is cancelled.

The mono mix control is like a balance control, and determines the amounts of left and right information that are summed. Under ideal circumstances, the left and right signals will be identical in level and phase. Old records are far from perfect, and many do not have identical characteristics in the two groove walls, even though they were recorded with a monaural cutter.

The best way to adjust the mono mix control is to put the lateral/vertical switch in the position just the opposite of the way the record was cut. Put in the vertical position for a lateral recording, and adjust the mono mix control for a null in the signal. A "perfect" record will yield complete cancellation, and only noise will remain. Then put the switch back in the lateral position for playback.

Most laterally-cut records have some vertical component, and vice-versa, so complete cancellation won't always occur—you simply adjust for the lowest signal level. Often old records have different levels of wear on the two groove walls. The mono mix control allows you to select either groove wall, or any mix of the two.

The EQS MK12 also has a 12-position gain control, allowing the preamp to be interfaced with a variety of consumer and professional equipment. K-A-B points out that this switch should not be used as a volume control. The EQS MK12 is intended to be used with a preamplifier, integrated amplifier, or other control center. Both unbalanced RCA and electronically-balanced ¼″ phone (tip-ring-sleeve) outputs are provided. The RCA jacks are high-end gold-plated, Teflon™-insulated types.

DESIGN PARTICULARS

Although schematics were not supplied with my review sample, a look inside revealed the general nature of the circuit topology (*Photo 3*). Each channel of the phono preamplifier uses switchable, passive equalization networks, situated between two gain modules, which are discrete and consist of five transistors each. One percent-metal-film resistors, plus Wima and Panasonic polypropylene capacitors, are used in the filter networks, and throughout the preamp.

The stereo fourth-order rumble filter is built around a National Semiconductor LF347BN quad J-FET op-amp. The line stage uses a pair of National LM837N low-noise, high-output-current, bipolar quad op amps—one of these is used for polarity inversion for the lateral/vertical switching, and the other is configured as a differential-output buffer amplifier. The output gain switch is not an attenuator—the ten-position switch is configured to change the gain of one of the line stage op-amp sections, by switching parallel feedback resistors. K-A-B also switches parallel capacitance, in order to maintain the same bandwidth at the different gain settings.

In a rather radical departure from today's norm in solid-state audio design, K-A-B has employed a single-ended power-supply topology. The supplied "wall-wart"-type power transformer is rated at 24V DC at 400mA, but even under the load of the preamp the transformer output actually measures 28V DC. Inside the preamp, this raw DC supply is fed to an L/C choke-input filter, and the filter output feeds an

LM317 pre-regulator with the output voltage set at +22V. The line stage and rumble filter op amps have their own regulators, a pair of 7818 three-terminal types. Each of the discrete gain modules in the phono preamplifier—four total for the two channels—has its own on-board 7818 regulator.

K-A-B uses the term "Polar Stable" to describe their single-ended approach. In an answer to an e-mail inquiry I posed, Barrett noted that "The polar design dictates that along the signal path there will always be a bias voltage present. I have come to regard this bias as having a stabilizing influence on the circuitry and components. I believe it plays an important role in the sound of my designs. Tube designs tend to share this similarity also."

In a single-polarity power-supply topology the signal path will, ideally, be at a potential equal to exactly half of the supply rail. For a variety of reasons, this is not always the case. If the signal path deviates from this ideal, asymmetrical clipping will occur causing a reduction in headroom (one side of the waveform will clip prematurely).

K-A-B has solved this problem by adding a fixed 9V bias, supplied by a 78L09 IC regulator. The bias holds the

TABLE 3
RECORDINGS USED FOR THIS REVIEW

RIAA STEREO LPS:

Wagner: *Die Walküre*—Ride of the Valkyries. Los Angeles Philharmonic Orchestra conducted by Erich Leinsdorf. Sheffield Labs Direct-to-Disc LAB-7 (Pressed in the US).

Rimsky-Korsakov: *Scheherazade*, Op. 35. Chicago Symphony Orchestra conducted by Fritz Reiner. Chesky RC4.

Wagner: *Der Ring des Nibelungen*, especially "Siegfried's Death and Funeral March" from *Götterdämmerung* (Side 11), and the "Forging Scene" from *Siegfried* (Side 3). Birgit Nilsson, Wolfgang Windgassen, et al. Vienna Philharmonic Orchestra conducted by Georg Solti. Decca 6.35500 (Mastered and pressed in Germany by Teldec).

Strauss: *Salome*, especially "Wird dir nicht bange, Tochter der Herodias" through Jokanaan's descent into the cistern (Side 2). Birgit Nilsson, Eberhard Wächter, et al. Vienna Philharmonic Orchestra conducted by Georg Solti. London OSA-1218 (Mastered and pressed in England by Decca).

Stravinsky: *Le Sacre du Printimps*. Cleveland Orchestra conducted by Lorin Maazel. Telarc DG 10054.

PRE-RIAA HISTORICAL RECORDINGS:

Schoenberg: *Gurrelieder*—"Lied der Waldtaube." Martha Lipton, mezzo-soprano. Philharmonic-Symphony Orchestra of New York conducted by Leopold Stokowski. Columbia ML-2140 (10″ LP; Rec. 1949).

Leoncavallo: *Pagliacci*—"Vesti la giubba." Giovanni Martinelli, tenor. Vitaphone Orchestra conducted by Hermann Heller. Vitaphone Soundtrack Disc, Matrix No. 300107 (33 1/3-rpm; vinyl pressing from the original metal part; Rec. 1926).

Cohan: *Over There*. Enrico Caruso, tenor. Victor 87294 (Acoustic; 78-rpm; "Wing" label; Rec. 1917).

Wagner: *Lohengrin*—"Mein lieber schwan!" Jacques Urlus, tenor. Edison Diamond Disc 83017-R (Acoustic; 80-rpm; vertically-cut; Rec. 1915).

Weber: *Oberon*—Overture. Berlin Philharmonic Orchestra conducted by Arthur Nikisch. HMV 1040 (78-rpm; acoustic; vinyl pressing from original G&T metal part issued by Symposium).

Beethoven: Symphony No. 7 in A Major. Philadelphia Symphony Orchestra conducted by Leopold Stokowski. Victor Set M-17 (78-rpm; "Scroll" label; Rec. 1927).

Strauss: *Also Sprach Zarathustra*. Boston Symphony Orchestra conducted by Serge Koussevitzky. Victor Set M-257 (78-rpm; "Scroll" Label; Rec. 1935).

Verdi: *Otello*—"Dio! mi potevi scagliar" (sung in German). Lauritz Melchior, tenor. New Symphony Orchestra conducted by John Barbirolli. HMV D2037 (78-rpm; British pressing; Rec. 1930).

Wagner: *Die Meistersinger von Nürnberg*—Act III Quintet. Elizabeth Schumann, Lauritz Melchior, Friedrich Schorr, Ben Williams, and Gladys Parr. London Symphony Orchestra conducted by John Barbirolli. Victor 7682 (78-rpm; "Scroll" label; Rec. 1931).

Beethoven: *Fidelio*—"Gott, welch' dunkel hier." Helge Roswaenge, tenor. Berlin State Opera Orchestra conducted by Bruno Seidler-Winkler. HMV D.B.4522 (78-rpm; British pressing; Rec. 1938).

Chopin: Ballade No. 4 in F-minor. Alfred Cortot, pianist. HMV D.B.7589-7590 (78-rpm; British pressing; Rec. 1933).

Wagner: *Lohengrin*—Prelude to Act I. Philharmonic-Symphony Orchestra of New York conducted by Arturo Toscanini. Victor Set M-308 (78-rpm; Rec. 1936).

signal path—specifically the outputs of the various amplification stages—exactly mid-way between ground and the supply rail at all times, ensuring symmetrical clipping and maximum headroom.

The EQS MK12 is capacitor-coupled, typically with Panasonic HFS electrolytics bypassed with polypropylene film capacitors. The main output coupling capacitors appear to have a triple-film bypass. The switching in a preamp with a single-ended power supply can produce clicks and pops if care is not taken to keep the outputs of coupling capacitors at 0V potential with resistive loading. All switches in the EQS MK12 are completely silent in their operation, including the much-used equalization selectors. The construction quality is excellent.

MEASUREMENTS

I made all measurements on my Sound Technology 1700B analyzer. *Figure 1* shows the frequency response in the *flat* position. The EQS MK12 circuitry has been designed for wide bandwidth, with the response at –2dB at 10Hz and –3dB at 100kHz. My measurement showed the –3dB point for the rumble filter to be at 28Hz, rolling off at 24dB/octave below that frequency.

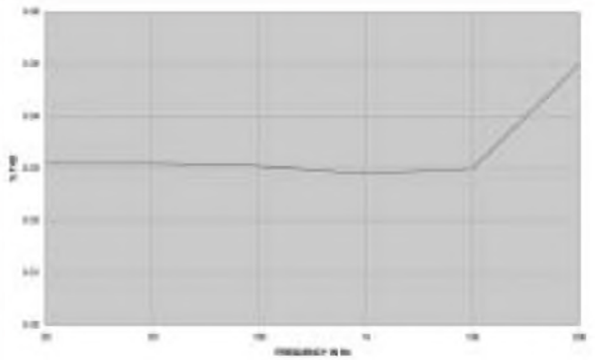

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FIGURE 5: Total harmonic distortion, measured in the flat position. THD remained around 0.03% across most of the spectrum.

I used the Jung/Lipshitz Inverse RIAA Network (*TAA* 1/80) to measure the RIAA accuracy, with 1V output at 1kHz as my 0dB reference. K-A-B does not specify the RIAA response—my measured results are shown in *Fig. 2*. K-A-B has built some infrasonic rolloff into the RIAA circuit, so the low end is at –0.7dB at 20Hz. This is a more conservative infrasonic rolloff than the IEC amendment to the RIAA specification. I have never liked the IEC call for a –3dB point of 20Hz (corresponding to a time constant of 7950µs). I have tried this in my own preamp and found that the degradation of the bass was quite audible.

K-A-B's solution is more sensible, putting the –3dB point nearly an octave lower than IEC. The RIAA performance is ideal throughout the critical midrange, measuring ±0.1dB from 500Hz to 20kHz, and ±0.25dB from 40Hz to 20kHz.

There are no inverse networks for the ten remaining curves included in the EQS MK12, so I simply measured the actual response of each one. I set my 0dB reference at 1kHz, with the preamp output driven to 0.5V out (corresponding to a phono input level of 5.3mV; the 1kHz gain of the preamp measures 39.5dB). I carefully monitored the preamp's output on an oscilloscope to make certain that the preamp was not clipping in the low frequencies.

The six 78-rpm curves are shown in *Fig. 3*, and the four 33 1/3-rpm curves are shown in *Fig. 4*. All measurements showed that the EQS MK12 performs as specified. When examining *Figs. 3* and *4*, bear in mind that I used 1kHz as a convenient reference for all ten curves. In *Fig. 3*, the E3, E5, and E7 curves show the K-A-B Fine Slope technique in the treble region, shelving at 10kHz.

K-A-B specifies f_3 for the LP curve as 50Hz, which made me initially suspicious. The original Columbia LP curve put the low-bass turnover at 100Hz. But, the low-frequency gain stop incorporated by K-A-B for the LP curve, combined with f_3 of 50Hz, produces the proper result, as *Fig. 4* shows. All frequency response measurements were essentially identical in both channels—any differences were beyond the resolving capability of my test equipment.

I measured total harmonic distortion in the *flat* position, with an input level of 11mV, which produced an output level of 1V. *Figure 5* shows THD to be around 0.03% across most of the spectrum, rising to 0.05% at 20kHz. Noise dominated the distortion products at lower frequencies, with some second harmonic introduced at 10kHz and 20kHz. There were no higher-order distortion components.

Changing to RIAA equalization lowered the THD to 0.01% at 1kHz, 10kHz, and 20kHz, with the distortion products consisting entirely of noise. Two-tone SMPTE IM distortion measured 0.011%. All distortion measurements were the same in both channels.

In the *flat* position, the output before clipping was 5.1V unbalanced, and 10.2V balanced. In the *flat* and RIAA positions, the maximum input level just before clipping was 62mV. Noise measured 71dB below 1V in the *flat* position with the input shorted; the high-frequency rolloff of the RIAA curve reduced the noise to 78dB below 1V.

THE SOUND

To evaluate the basic sonic quality of the EQS MK12, I used several of my reference stereo LPs (*Table 3*), all cut with RIAA equalization. My reference LP

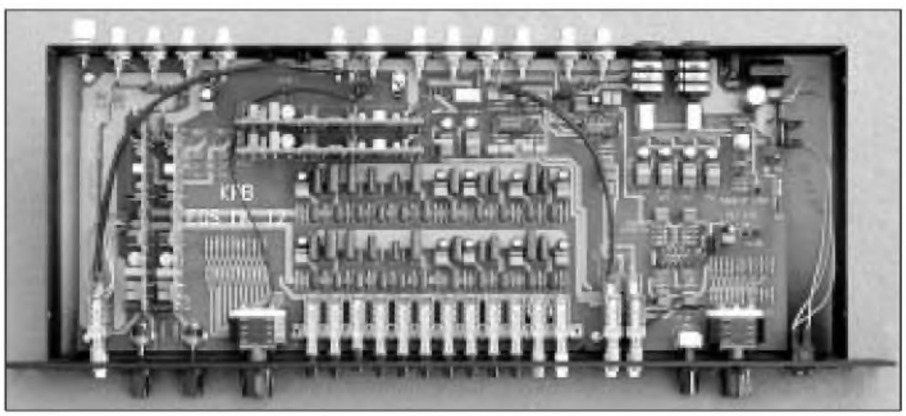


PHOTO 3: Inside the EQS MK12. The vertical PC boards are the discrete gain blocks. In the center are the passive equalization circuits.

PHOTO 4: Putting the EQS MK12 through its paces, with 78s (left) and LPs (right).

playback system is my own belt-driven, AR/Merrill-based turntable fitted with a Grado Signature tonearm and an Adcom XC-MRII high-output moving-coil cartridge. The turntable is powered by an electronic speed control of my own design (*Photo 4*).[4]

The EQS MK12 proved to be a fine-sounding performer offering a warm and detailed sonic presentation. The treble region is silky and smooth, lending itself to fatigue-free long-term listening. Overall, I found the warmth and liquidity in the sound to be somewhat tube-like, though not over-ripe. The preamp offers a hint of the euphonic qualities of a good tube preamp, but it is not overly colored.

Soundstaging is a bit narrower and shallower than my reference preamplifier (my extensively modified Adcom GFP-565), but localization is generally very good. I had no difficulty following the subtle stage movements of the singers in the Culshaw-produced Wagner and Strauss recordings conducted by Georg Solti. The bass region is a bit reticent, lacking the weight and impact of my reference preamp, but it is clean and well-defined. Overall, the EQS MK12 is a solid performer with stereophonic/high-fidelity material, offering satisfying musical performance.

Over the course of several months of listening, I played literally dozens of pre-RIAA recordings, mostly 78-rpm discs, but also including early 33 1/3-rpm material. My 78-rpm playback system includes a Technics SP-15 turntable and an SME 3012R tonearm mounted on a custom base with isolation feet, plus a Stanton 500A-series cartridge with a variety of truncated stylii (*Photo 4*). I also use this turntable for some 33 1/3-rpm recordings, including 16″ transcriptions, Vitaphone soundtracks, and early-1930s Victor LPs.

Table 3 contains favorite recordings that I found especially useful, but it is only a partial list. The EQS MK12 did an excellent job with the discs I auditioned. Overall, the EQS MK12 easily outperformed my modified McIntosh C-8, which has been my reference 78-rpm preamp for over 15 years.[5] The EQS MK12 offers a cleaner and more detailed sound than the C-8, with a more open and transparent treble region.

I found the EQS MK12's curves to be

intelligently chosen, with the AC curve offering just the right amount of warmth for most well-recorded acoustic discs. The moderate high-frequency rolloff reduces surface noise without cutting into the limited treble region on acoustical recordings. It is rare to find an electrically-recorded disc that won't offer musically satisfying performance with one of the supplied curves. If you've never heard 78s played with proper equalization, the EQS MK12 is likely to be a revelation.

The lateral/vertical switching and mono mix controls worked superbly. A proper mix can make a great difference in the sound of 78s, and I consider this control essential for optimum playback. There were times when I wished that the rumble filter began at a higher frequency. Many 78s have audible rumblings in the 40-60Hz region, including the Meistersinger Quintet recording on my list. A higher corner frequency would admittedly compromise many recordings, so I really can't fault the designer's choice. Serious collectors will probably find a parametric equalizer an essential accessory.

It is often desirable to put the processor loop after the lateral/vertical switching and mono mix. If a stereo equalizer is used ahead of these controls, you can't get a proper null with the mono mix controls unless the two channels of the equalizer are precisely matched.

A better solution is to put the equalizer at the output of the EQS MK12. In this case, you need only a mono equalizer, which can then feed your control preamp. I suggest the Behringer Ultra-Q® Pro PEQ2200 as a cost-effective solution, or the Symetrix 551E for greater flexibility (K-A-B sells the Symetrix, and most pro-audio dealers carry both).

I know a number of professionals involved in commercial transfer of historical recordings, and many insist on separate control of the bass turnover and treble rolloff frequencies, which the K-A-B doesn't offer. There are many electrically-recorded 78s with 6dB/octave high-frequency preemphasis curves that are not exactly complemented by the "Fine Slope" rolloff in the EQS MK12, though I found that the "Fine Slope" curve worked well with most of the 78s I played. With the addition of a parametric equalizer, however, you can make sufficient adjustment to accommodate a variety of 78-rpm records. The AES curve, with its 6dB/octave slope, is also useful on many 78-rpm discs.

CONCLUSION

The K-A-B EQS MK12 is an excellent solution to the problems facing collectors of historical recordings. A high-quality preamplifier with a variety of equalization curves will breathe new life into old recordings, and the EQS MK12 is probably the best all-around product of its kind currently available.[6] Complemented with a good turntable, cartridge, and stylii, the EQS MK12 makes a fine playback system for 78-rpm and other vintage material. The AC and FLAT settings should also work well with cylinders.[7]

The EQS MK12 comes with a helpful instruction booklet and a chart offering recommended equalizer settings for a wide variety of recordings. If you are a serious collector, the EQS MK12 deserves serious consideration.

Manufacturer's response:

Thank you for allowing me to comment on this review. I was hesitant at first to consent to this review until Mr. Galo informed me of his experience with historic recordings. His reputation in the audiophile world is well known. His knowledge of historic recordings and playback requirements make this review all the more complete and informative. This review should be required for all hi-fi equipment reviewers.

I am particularly pleased with the measurements. The other audiophile magazines have ceased to measure phono stages. All of our EQ networks are mathematically derived. The measurements show the accuracy of this technique. Measuring each EQ component to a tighter tolerance would produce more accuracy in the curves. However, it is not likely to be an audible improvement.

One of several measurements we make on the individual transistors that make up the gain stage is very low frequency noise. The output devices used in the gain stage can exhibit "popcorn" noise. This is a noise burst of very short duration. I discard devices that exhibit this phenomenon. Feedback loops should not have to deal with this type of noise. Even though in a feedback design the noise would be effectively "hidden" by the very action of the feedback loop. It is a variable I can stabilize by screening out the noisy devices.

To maintain the lowest possible output impedance the EQS MK12 features a relay switch that "short circuits" the outputs for about 15 seconds at power up. This does two things: It prevents turn on thumps and it forces a very complete charging of the output capacitors. This reduces the capacitor ESR (Equivalent Series Resistance) to its lowest level. It also minimizes leakage currents.

The reason for placing the process loop before the mono mix is to have an output just off of the second class A gain stage. The Send #2 output can be used for hi end stereo playback and the entire process stage is precluded from the signal path. Though the gain is fixed at 36dB, it is a very pure signal path.

One change we have made to current production units is to decrease the overall gain by 3dB. This increases the overload margin from 62mV to about 91mV.

The EQS MK12 can be found in countless college libraries and sound archives around the country including the US Library of Congress.

An RIAA only version is also available as the EQS MK2.

Kevin A. Barrett

REFERENCES

1. Lipshitz, Stanley, "On RIAA Equalization Networks," *Journal* of the Audio Engineering Society, June 1979.

2. Galo, Gary, "Disc Recording Equalization Demystified," *The LP is Back!*, Peterborough, NH, Audio Amateur Press, 1999 (available from Old Colony). The article is based on a paper given at the Association for Recorded Sound Collections conference in 1996, originally published in the Fall 1996 issue of the ARSC Journal.

3. RIAA has a website, www.riaa.org, but you won't find the RIAA LP curve here, just a link to National Semiconductor's datasheet and audio application note for the LM833 opamp.

4. Galo, Gary, "AR System Drives New Turntable," *Audio Amateur* 3/85; "An Electronic Speed Control," *Audio Amateur* 1/86; and "The Belt-Driven Turntable Revisited," *Audio Amateur* 3/88. These articles have been reprinted in *The LP is Back!*.

5. Galo, Gary, "A Preamp for Vintage 78s," *Audio Amateur* 1/85. Reprinted in *The LP is Back!*. My C-8 has been modified beyond what was described in this article.

6. The FM Acoustics Resolution 222 features separate control of bass- and treble-turnover frequencies, but the lack of a flat position and mono mix control makes it less useful for 78-rpm discs, and its $18,500 price tag puts it out of the reach of all but the affluent.

7. There's very little equipment available for electronic playback of cylinders. A couple of interesting products can be found on the website of Nauck's Vintage Records at www.78rpm.com. Their Advanced Cylinder Technology Reproducer houses a Stanton 500 cartridge and can be fitted to a number of original Edison cylinder phonographs.

Product Review

DH Labs Cables

By Gary Galo

D.H. Labs Air Matrix Interconnect; Q-10 Loudspeaker Cable; Deluxe Toslink Optical Digital Cable; Improved D-75 S/PDIF Digital Interconnect. D.H. Labs, Inc., 612 N. Orange Ave., Suite A-2, Jupiter, FL 33458, 561-745-6406, www.silversonic.com. Air Matrix, $195/1m pair; Q-10, $225/8ft stereo pair, $10/ft bulk; Deluxe Toslink, $45/1m; D-75 RCA, $75/1m; D-75 BNC, $100/1m.

PHOTO 1: D.H. Labs newest cable products. From top to bottom, the improved D-75 digital interconnect with silver-plated center conductor, Deluxe Toslink Optical Cable, Q-10 Speaker Cable, and Air Matrix analog interconnect fitted with a locking RCA connector.

Back in *Audio Amateur* 2/94 I reviewed some high-value cables from D.H. Labs, the Silver Sonic BL-1 interconnect and T-14 loudspeaker cable. Since that time, D.H. Labs has developed some impressive new cables for their Silver Sonic line, designed to compete with some of the most expensive cables available. The Air Matrix Interconnect is manufactured with oxygen-free copper plated with high-purity silver, and a Teflon-foam dielectric.

The Air Matrix dielectric is a sophisticated PTFE Teflon foam that is actually 60% air. Unlike conventional foam dielectrics, the Air Matrix is not simply a solid plastic containing air bubbles. This dielectric has a texture resembling a fine matrix, and can only be seen under a microscope. The excellent uniformity of the Air Matrix dielectric gives it excellent transmission properties to beyond 17GHz.

D.H. Labs' Darren Hovsepian explained that a 60% air/40% Teflon ratio yields an optimum combination of rigidity and performance. Once you increase the air component to 70% or beyond, the dielectric becomes too soft, and is too easily compressed. Compared to solid Teflon—such as that used in the BL-1 interconnect—the Air Matrix exhibits both lower capacitance and lower inductance. The dielectric constant of the Air Matrix is 1.4, compared to 2.1 for solid Teflon, and the propagation velocity is 84%, compared to 69% for Teflon (with dielectric constant, the lower the number the better; propagation velocity is just the opposite).

The Air Matrix cable is around 5/16″ in diameter, and is not sold in bulk form. D.H. Labs offers the cable in a variety of stock lengths, and will assemble any custom length required. Check their website for pricing.

D.H. Labs tells me that some individuals have been buying BL-1 cable bulk, and selling it with inferior RCA connectors, claiming that these products are still D.H. Labs cables. In order to guarantee quality control over this new cable, the manufacturer will only sell it pre-fitted with their premium-quality gold-plated, Teflon-insulated locking RCA connectors. These are among the very finest RCA plugs you'll ever see, and a distinct improvement over my old standby, the Canare F-10.

Both the shield and center-pin are a one-piece design from front-to-back. These connectors also feature a highly effective clamping system. After plugging the connector into an RCA jack, you simply turn the threaded outer sleeve to tighten the shield connection. The locking RCA connectors are also available separately for $15 each.

AIR MATRIX PERFORMANCE

D.H. Labs sent me enough Air Matrix cable to connect a 1m pair between my D/A converter and preamp, and a considerably longer run (about 15 feet) to go between my preamplifier and power amps. Previously, I had been using identical lengths of BL-1 cable. I found the Air Matrix cable to be a remarkable improvement over the BL-1, not at all subtle.

The most noticeable difference is the increase in fine detail and resolution, combined with greater air and space around the instruments. The treble region is utterly smooth and transparent, and the tonal balance is extremely neutral. The increased resolving power of this cable improves soundstage localization, as well, resulting in a more lifelike 3-dimensional sonic presentation.

The Air Matrix cable is not inexpensive, but it is well worth the asking price when used in a high-performance audio system. It should compete very favorably with cables costing hundreds of dollars more in high-end retail stores. This is a truly remarkable high-end cable.

Q-10 LOUDSPEAKER CABLE

The Q-10 loudspeaker cable is based on the same materials and construction as the T-14 I reviewed back in 1994, including silver-plated, oxygen-free copper conductors and pure Teflon insulation. But, the Q-10 is a four-conductor configuration—two 14AWG conductors and two 12 AWG in a single jacket. Used in parallel, this yields an equivalent 10AWG cable. Alternately, you can use one length for bi-wiring, with the larger conductors on the woofers, and the smaller ones on the higher frequency drivers.

D.H. Labs sent me enough wire to go "all-out." Four lengths of Q-10, two per channel, feed my ACI Sapphire III satellite loudspeakers, in a bi-wired configuration (with my custom, all-polypropylene crossover on the Sapphire III tweeters), and two more short lengths feed the ACI Sub-1 subwoofers. A pair of Monarchy SE-100 Delux power amplifiers power the loudspeaker systems.

The Q-10 cable was designed primarily for difficult loudspeaker loads, so I was unsure whether there would be significant differences between the new wire and the T-14 on my ACI system, which is a rather benign 4Ω load. I was surprised to find a noticeable improvement in the spatial characteristics of my system with the Q-10 cable. Localization within the soundstage is more precise, with a more realistic sense of the acoustic of the recording venue. The lower midrange has greater warmth and harmonic richness, particularly evident on massed low strings.

If you use only one length of Q-10 in a bi-wired setup, I would expect the performance to be quite similar to two runs of T-14. I recommend double runs of the Q-10 if you can afford it. The Q-10 didn't provide much improvement in the low bass, compared to the already excellent T-14. But, my subwoofer cables are less than two feet long–I would expect a greater improvement with longer cable runs.

D.H. Labs sells a variety of gold-plated spade and banana connectors for their speaker cables. For 10-14 AWG cables, they have the SP-1 for normal-size 5-way binding posts, and the SP-1W for wider posts. The SP-2 and SP-2W will accept an 8 AWG cable. Two conductors of D.H. Labs Q-10 will fit the SP-1 or SP-1W.

The SP-2 and SP-2W will accept four conductors, though you may need to enlarge the opening slightly. You'll need the SP-2 variety for the power amp end if you are bi-amping using two separate runs of Q-10 cable. I strongly suggest soldering rather than crimping these connectors (or soldering *in addition* to crimping).

D-75 AND TOSLINK IMPROVEMENTS

D.H. Labs has made a slight change in their D-75 S/PDIF digital interconnect–the solid center conductor is now silver plated. I compared digital cable using my Parts Connection D2D-1 Sample Rate Converter and DAC 3.0 Digital Processor. I fitted a 1m length of the improved cable with Canare 75Ω BNC connectors and found it to be slightly better than the original D-75. The sonic presentation is a bit more transparent and detailed than with the previous D-75.

This is an incremental improvement, and you don't need to throw out your old D-75. But, on a high-resolution system, the difference is audible. Darren Hovsepian says that he is also receiving positive reports on the video performance of the improved D-75 in high-end home-theater applications.

D.H. Labs is also carrying a Deluxe Toslink Optical cable, which I found markedly superior to the Kimber OPT1 that I reviewed in *Audio Electronics* 4/99. For comparison, I gave these cables the ultimate test: passing 96kHz/24-bit data from my Pioneer DV-525 DVD player. I realize that Toslink was not designed for 96kHz operation, but when the DV-525's digital output is set to 96kHz, *both* the S/PDIF coax and the Toslink optical output operate at this frequency.

On the Classic Records DVD transfer of the Vox/Turnabout's 1967 recording of Rachmaninoff's *Symphonic Dances*, the strings sound quite gritty and unrefined with the Kimber interconnect. Changing to the D.H. Labs Deluxe Toslink cable removed most of the high-frequency grunge. In fact, I was quite surprised at just how good this recording sounded using the D.H. Labs cable–not as good as their D-75 coax fitted with 75Ω BNC plugs, but very respectable nonetheless.

Darren Hovsepian said that they simply used the widest bandwidth optical fiber they could find for this cable, but I noticed two other differences. The termination on the D.H. Labs cable has a visibly better polish than the Kimber, and the D.H. Labs cable also makes a mechanically tighter fit when plugged into the Toslink transmitter and receiver modules. The Kimber interface is looser, and easily wiggled out of optimum alignment. Imprecise mechanical alignment has always been a shortcoming of the Toslink interface, but the D.H. Labs cable improves the situation considerably.

Another nice feature of the D.H. Labs cable is the protective end cap. As you can see in *Photo 1*, the cable has a retainer which prevents you from losing the end cap, which will help prevent damage to the cable terminations during storage. The D.H. Labs Deluxe Toslink Optical cable is also priced $15 less than the Kimber.

D.H. Labs' new cable lineup, especially the exceptional Air Matrix analog interconnect, has put them in a new league among cable manufacturers. Their earlier cables provided great performance for the money, and I found them eminently satisfying in my own reference system. These new cables put D.H. Labs in direct competition with some of the most expensive cables available, and the prices are still very fair considering the high level of performance they offer. You won't be disappointed with these excellent products. ❖

Book Review

Handbook For Sound Engineers, 3rd Ed.

By Richard Honeycutt

Published by Focal Press, 225 Wildwood Ave., Woburn, MA 01801-2041, www.focalpress.com. Available through Old Colony Sound Lab (BKS28), PO Box 876, Peterborough, NH 03458-0876, 603-924-6371, toll-free (US/Canada) 888-924-9465, FAX 603-924-9467, custserv@audioXpress.com, www.audioXpress.com, $120.

The old-timers among the readership will remember Howard Tremaine's *Audio Cyclopedia*, a 3-inch-thick compendium of questions and answers from all fields of audio. My copy still graces the bookshelf over my bed, in case I should awaken some night wondering about the specifics of the RIAA curve.

When the first edition of the *Handbook for Sound Engineers* was published in 1987, it was subtitled *The New Audio Cyclopedia*, in order to identify its encyclopedic content with the earlier work. With the recently-published 3rd edition, the encyclopedic approach remains, but the subtitle has finally been dropped.

Glen Ballou remains the editor of this, as of the first two editions. Glen has been active in sound system design and installation since 1970, and is an active member of the AES, SMPTE, and IEA. He is also a former governor of the AES.

The book is arranged into 47 chapters organized into 7 parts: Acoustics, Electronic Components, Electroacoustic Devices, Electronic Audio Circuits and Equipment, Recording and Playback, Design Applications, and Measurements. The depth of coverage of topics is far beyond the level of most general audio reference books. For example, the chapter on auditorium and concert-hall acoustics covers not only the common basics such as reverberation time and reflecting surfaces, but many new metrics such as bass ratio, strength *G*, interaural cross-correlation coefficient *IACC*, and center time t_s.

The microphone chapter begins with a discussion of various polar patterns, but progresses to distinguish among single-entry, three-entry, and multiple-entry cardioid designs. Application and performance of rifle, shotgun, parabolic, X-Y stereo, ORTF, and other microphones are presented. The chapters on sound-system design and computer-aided sound-system design provide excellent background information.

One of the great strengths of this reference book is the choice of eminent authors, each an acknowledged master in his area of audio specialization. Thus Dr. Wolfgang Ahnert, developer of the EASE electroacoustic modeling system, writes on concert-hall acoustics and computer-aided sound-system design. Jay Mitchell of Frazier authors the chapter on loudspeakers. Joe Hull of Dolby Labs describes surround sound. The list goes on.

Another strength of the book is the provision of generous end-of-chapter references.

Perhaps readers of this publication will be most interested in the two chapters on loudspeakers. In the first, Jay Mitchell not only gives the qualitative discussion of how speakers work, but he also compares design features, such as Peavey's focused-field geometry and JBL's symmetric-field geometry. He also discusses electrodynamic, electrostatic, and piezoelectric speakers in some detail. Heat-dissipation considerations are addressed, especially as connected to the geometry of motor design, and there is an overview of horns.

Mitchell lays out design considerations for crossover networks, discusses speaker specifications, and introduces theory (but not the practice) of enclosure design. And finally, there is an overview of loudspeaker measurement, including T/S parameters.

Ralph Heinz authored the chapter on loudspeaker cluster design. After laying out the problems inherent in the art of cluster design, he stresses a proprietary technology called the True Array Principle.

Also covered in this book are some topics of relatively recent interest, such as in-ear monitoring (Gino Sigismondi of Shure Application Engineering) and Assistive Listening Systems (Glen Ballou).

From my perspective, there is one significant omission in the coverage provided by the *Handbook*. Over the past 50 or so years, there has been a slow development of loudspeaker line arrays. Today, this type of array has advanced to the point where it is of real importance in sound system design.

Line arrays fall into two categories: groups of speaker enclosures that are placed so as to function as an array in concert venues, and single-enclosure line array systems for auditorium and worship-center applications. The behavior and application of such arrays is in some ways counter-intuitive. A chapter devoted to such arrays would have been a very valuable addition to the book.

Much of the book has been rewritten since the last edition, bringing the information up-to-date. The presentation is clear and concise, with the level of mathematics varying from author to author: from many chapters containing almost no math to a few utilizing integral calculus. The presence of 1000 illustrations helps also.

You must recognize that the purpose of any general reference book is not to serve as a textbook, but to help practicing professionals keep up-to-date in their fields, and to help them access relevant information in related fields. *Handbook for Sound Engineers* achieves this purpose well. In fact, it presents a level of detail not often found in general references. Practicing audio professionals–both novice and experienced–and serious amateurs in the field of audio will find this book a great value. ❖

Glass Shard

A Grounded Grid Line Stage

I built this line stage with reference to Dirk Wright's article, "A Grounded Grid Line Stage" (*GA* 5/95). I changed some resistor values for better performance according to my listening test. I biased +120V to the V3's (12AU7) cathode to lower the cathode and filament potential; therefore, two separate filament windings are necessary (*Figs. 1* and *2*). You cannot plug in V3 alone, as high voltage will cause positive bias!

I derived the regulated power-supply circuit from Sound Valve's VTP 100 (*Fig. 3*). I have found the LM 7812 temperature is very high, although, after two hours, no failure has occurred. TIP 50 heatsink will carry high voltage if not well isolated (*Fig. 4*). Be careful, as high voltage will still exist a few minutes after power has been switched off. A short circuit will burn all transistors. ❖

Ignatius Chen
Bandung, Indonesia

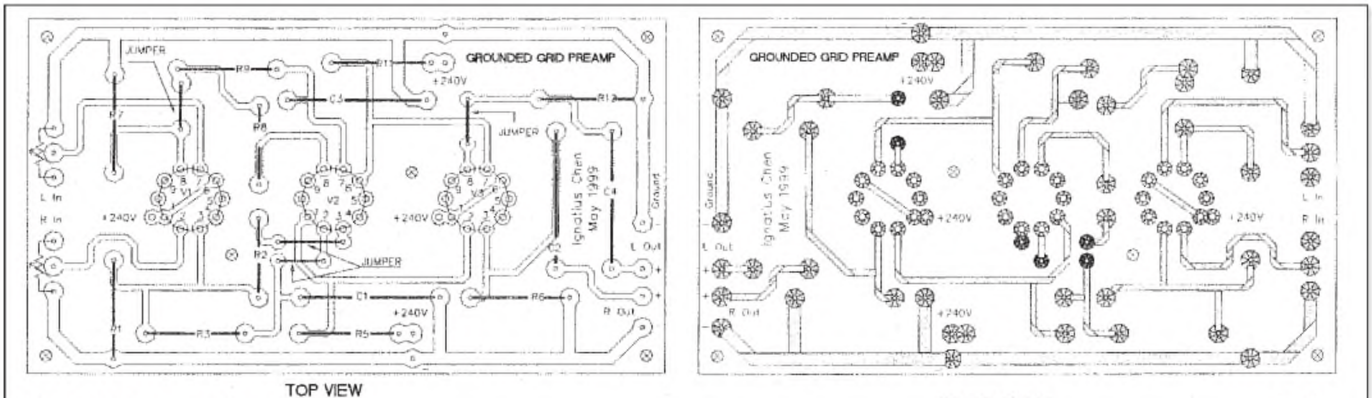


FIGURE 1: Preamp pc boards (75%).

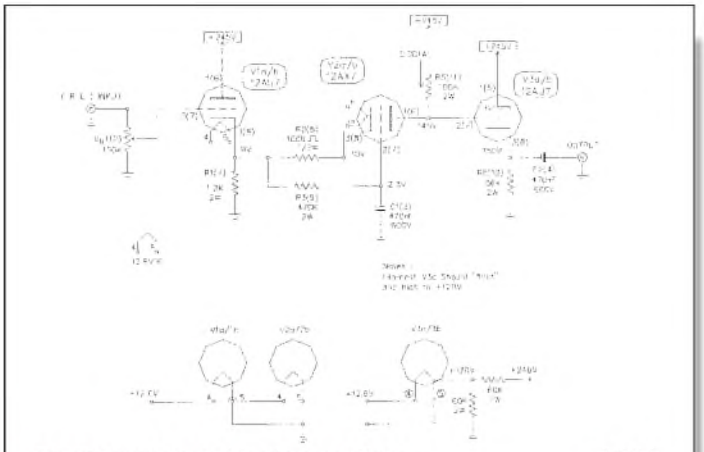


FIGURE 2: Grounded grid preamplifier. G-1649-2

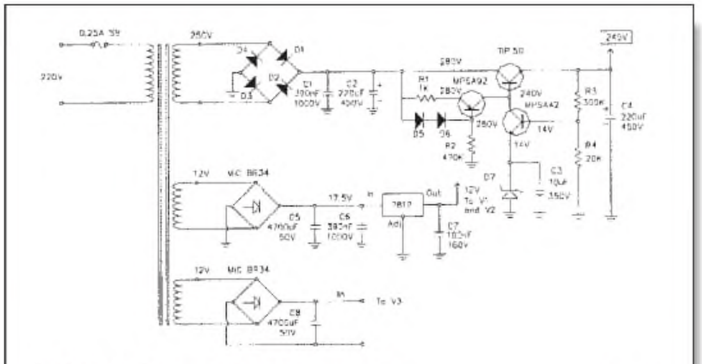


FIGURE 3: Preamplifier power supply. G-1649-3

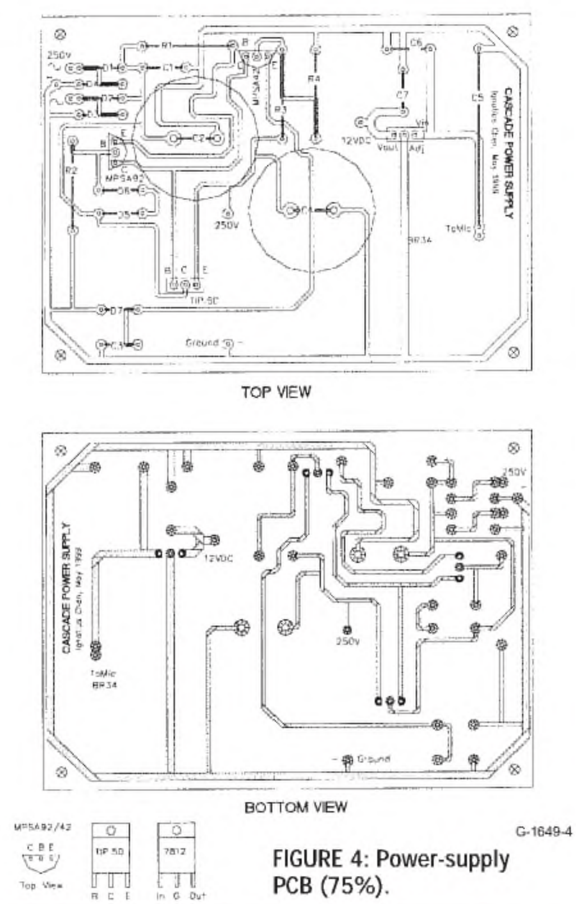


FIGURE 4: Power-supply PCB (75%).

Xpress Mail

FORMAT CORRECTION

The circuit I sent you and published in April '02 *aX* ("I^{2}S Follow-Up," p. 66) won't work as printed. This is partly my fault for not examining the timing diagrams more carefully, but also Dr. Rubinson's. What he calls the Sony-Burr-Brown format is not what those companies use. The Sony-BB format is right justified.

For this circuit to work you need left-justified format. This, in conjunction with the flip-flop and inverter, should give an effective I^{2}S input 16–24 bits without changing any switches on the board.

Now the input word length DIY switches disappear, and pin 3 is low, pin 4 is high, and pin 5 is high.

Also, the inverter and flip-flop need pull-up resistors (4k7?). The LS receiver device needs a little help driving the CMOS devices. Wouldn't hurt to add them to the X TIN, DATA, and DEM lines going directly to the filter as well.

DF 1704
Pins

1	DATA IN	
2	BCKIN	
3	L	(replaces
4	H	input DIP
5	H	select switcher)
6	XTIN	(master clock)
7	NC	
8	PS Ground	
9	NC	
10	L	(hardware mode)
11	L	
12	H	
13	NC	
14	H	
15	L	
16	DEM	
17	L	(sample rate for
18	L	DEM 44.1kHz)
19		(selectable output word
20		length with switches)
21	NC	
22	V_{DD} +5V	
23	DOR	
24	DOL	
25	WCKO	
26	BCKO	
27	L	(filter rolloff for 44.1kHz data)
28	WCLK IN	

Rick Bergman
Missoula, Mont.

LIGHTNING STRIKES TWICE

Usually, I find that magazines and magazine subscriptions are a waste of my time and money. Mainly, the lurid covers are designed to get my attention, and then the contents are worthless. Many a periodical has made a quick trip to the recycle bin at my house. At least, I found that your publication entertained me, challenged my mind and scientific reasoning, and delivered as promised, if nothing more.

Mr. Erath's article, "Analytical Horn System," in the March '02 issue really changed that. In two paragraphs he totally convinced me of the value of single-ended amplifier design when he pointed out that a really efficient loudspeaker magnifies the crossover distortion inherent in push-pull designs.

Let me make two points here:

1. I have never been a fan of horn loudspeakers.
2. It is truly a telling irony that a speaker article makes clear a longstanding problem with amplifiers.

Naturally, I have heard of crossover distortion, but I did not know much of the subject. (In my own crude tests, some very low efficiency design speakers appeared to have a superior sound—with a simple receiver source—over some more ordinary configurations.

This could be from the above, or due to the "tailoring" of the curve, using selective efficiency.)

I would stop there, but lightning struck twice, as it were. Graham Maynard's article on sub-bass equalization (p. 38) really made clear the flaws in woofers, and bass in general.

Think about it: the "electronics" article made clear the flaws in speakers; and the "speaker" article clarified the shortcomings of electronics! Double irony!

From that we can draw a few good conclusions:

- The above reinforces the concept of the system as a whole—with interaction between parts.
- These two authors both have legitimate, although vastly different, approaches, showing that there is room and a need for them.
- Both amplifiers (and other electronics) and speakers need a lot of developmental work in the future.

For myself, now I really would prefer a single-ended amplifier (when I can afford it). And now I'm really rethinking that fabrication of a 30″ woofer project I've been contemplating.

Nathan Shinder
Levittown, N.Y.

I note with great interest Louis Erath's article on a horn system using an Altec 805B horn and 288-8K driver (March '02, p. 18). Unfortunately, these components are rather difficult to obtain, as Mr. Erath notes. I wonder whether other more common speakers may be pressed into service. The requirement seems to be for a fairly narrow dispersion horn and a high power, large diameter, woofer.

Some Tannoy drivers such as the 15″ dual concentric K3838 might fit the bill. This concentric 2-way with a horn above 1,000Hz has 90° vertical and horizontal dispersion for –6dB at 10kHz. Not as narrow as the Altec, but perhaps still suitable for the concept.

The K3838's woofer section has sensitivity 92dB/1W/m, f_s 22Hz, Qe 0.19 Q_{TS} 0.18, V_{AS} 483 liters. It has a thermal power rating of 120W and can handle peaks of 500W. Its paper cone is made more rigid with paper rib re-inforcement. It seems to me that you could make this also operate satisfactorily, using Mr. Erath's patented ES woofer design.

I modeled a sealed box for it and found a very small 34-liter box would have a system resonance of 85Hz and that 20dB boost would provide 20Hz @ 90dB from a 100W amp—this being about the maximum. Over 100dB is available from 30Hz and up. In the non-boost area this speaker can produce 121dB peaks if 500W is available, so it is no wimp. You could perhaps use a slightly larger box to bring the resonance down a bit and require less boost.

Unfortunately, I didn't gain enough understanding from the details published to know how to design or build an ES system. I would be very interested to know whether Mr. Erath thinks my idea has any merit and, if so, to have further details of his ES system made available. I think further details may be required even to attempt to construct a speaker as described. Figure 2 mentions an Appendix A, which may shed some light, but this was not published.

There may be quite a few Tannoy enthusiasts who would love to attain very low bass as well as retain the delicacy and refinement possible from using an SET amp above 1,000Hz and with the huge bonus of a far more domestically acceptable enclosure size.

I fear many Tannoys may have been retired due to their love of huge enclosures and that this idea may liberate them for a new lease on life. I believe they are capable of the analytical focus of which Mr. Erath speaks, if well positioned and partnered with the right equipment.

Thanks for an inspiring article, but more details, please.

Bart Shepherd
Epping NSW Australia

Louis Erath responds:

Appendix A in my original manuscript is labeled Fig. 1 in the article, however, we neglected to remove the reference to it in Fig. 2. audioXpress published all of the information which I submitted. I will, however, provide any additional details which may be required to build an ES woofer.

In Fig. 2 of the article, T1 is the most im-

portant element. I used a powered permalloy toroidal core of a type originally developed for telephone filter circuits. This core is 1.57″ in diameter and has a mu of 125. There is probably an Australian equivalent. However, a core made of permalloy or mu-metal laminations, butt-stacked with an air gap, will probably work as well.

The secondary winding goes on first and has 200 turns of #24 enameled copper transformer wire. This yields an inductance of 6.6mH. The primary winding has 20 turns of #14 transformer wire for a turns ratio of 1:10. The resistor shunted across the secondary is 20Ω (1 or 2W metal film or wirewound). This reflects 0.2Ω into the primary which is in series with the 8Ω woofer voice coil. L1 has 600mH of inductance with a series DC resistance of 175Ω.

This choke in my ES woofer measures 1.25″ × 1.5″ with .5″ butt-stacked laminations plus an air gap. It is tuned by C1 to the resonance frequency of the woofer in its enclosure.

The symbolic amplifier in Fig. 2 shows the signal being fed to the non-inverting input and the feedback applied to the inverting input. The feeback is fed through a 50 to 100µF non-polarized electrolytic to a shunt resistor of about 50Ω. The amplifier has a typical sensitivity of about .7V RMS input for 100W output.

I use an electronic 500Hz low-pass crossover network; however, a passive network should work just as well if the driving source is of sufficiently low impedance. If L1 and C1 produce a dip in the response at 75Hz, a resistor may be used in series to flatten the response.

EXCELLENT

Just a quick note to say that T. D. Yeago's article ("A Sensible McIntosh 225 Refurbish," May '02, p. 52) is one of the best that I have seen in years concerning the McIntosh 225. The author has done an excellent job! The 7591 pentode has not been my favorite choice of output tubes due to screen dissipation problems (grids not aligned) and the fact that it doesn't use a wound suppressor grid, like the EL34.

The tubes do sound fine in the McIntosh circuit, but using a true beam-power tube such as the 5881 should really improve the McIntosh 225 (more efficiency), as well as the soft-on, soft-off heater supply. The adapter plate he has designed really does the trick and does not take away from the appearance of the amplifier.

An excellent article; well-written and engineered.

Joseph K. Risher
Sounds Great! Enterprises
Stone Mountain, Ga.

PHONO PREAMP

First, thanks to Eric Barbour for providing some of the most consistently interesting and innovative audio designs anywhere. I used the constant-current sources (CCS) from his pocket preamp project (*GA* 4/99) in a 6ER5/6GK5 phono project and am very pleased with the results. As long as they're carefully matched, the LND150s can be paralleled to provide a tiny, high-voltage CCS from 1–5mA, and with only one or two parts, in contrast to most of the CCSs appearing on the internet lately which range from 8–20 parts and require hundreds of extra volts. When you're trying to supply four or even six tubes with their own CCSs, this is a huge advantage.

As one who's been fascinated by the 416B since the late '50s, I was especially interested in the Planar-Triode Phono Preamp (Nov. '01, p. 18). Mr. Barbour's response to Allen Wright's concerns ("Xpress Mail," April '02) was fairly reassuring; I was particularly nervous about current through the cartridge. I am also wondering about input capacitance: with Miller effect couldn't it reach 300–400pF and cause high-frequency rolloff? I am planning a 416B/417A phono stage using contact bias on both stages; this has been inspired mainly by Mr. Barbour's articles.

On the same topic, Robert Gomes asked about 7077 availability ("Xpress Mail," April '02). tubesontheweb.com offers them for $58 new and $29 used. I bought some used ones on eBay cheap last month, so I'll probably try them in the preamp, too. Both they and the 6S17K-V sound much easier to use than the 416 because they don't run nearly as hot.

Fred Humphrey
Campden, Ontario

HIGH COST OF DRIVERS

Almost every recent issue seems to have a speaker construction article featuring expensive drivers. No doubt some of them are good, but it's been my experience that most of them are not. It's not difficult to imagine a newcomer to our avocation, working hard and saving his pennies to buy expensive drivers he's never heard. I've been down this road more than once, and have been disappointed virtually every time. Even though there are a lot of intelligent and skilled people manufacturing drive units, it's been painfully obvious to my ears and wallet that most of them are lacking some fundamental intuition about what makes a speaker sound like music.

One recurrent problem is harshness in tweeters. I remember a reviewer's opinion that they were "spitty little resonators." Not far from the mark, I'd say. If you spent $900 for a pair of tweeters, you'd expect them to have zero harshness, wouldn't you? Sadly, I can report this is just not the case.

About a year ago I bought a pair of MCM 53-630 inverted dome tweeters as an experiment. They cost about $15 each. Much to my surprise, they were less harsh than *any* cone or dome tweeter I tried. I'm not going to name names, because you wouldn't believe me, but if you look through '99 and '00 *Speaker Builder* issues, you can see all the ones I didn't like and sold in Yard Sale.

The 53-630s were cheap enough that I could experiment with them. They can be improved by applying a "ring" of Pliobond on the inverted dome directly above where the former joins underneath. The diaphragm is about 1″ in diameter, and the voice coil diameter is ¾″.

A warning here: Pliobond has *nasty* fumes. I only use it outdoors on a windy day. It will dissolve a nylon brush. You need a little horsehair brush (try your local hobby shop) to apply it. This mod further reduces harshness.

The radiation pattern meshes nicely with a cone woofer, but the front mounting plate needs some damping. I used a layer of Microsorb I had lying around, but I'll bet felt would work as well.

Rick Bergman
Missoula, Mont.

PANEL DAMPING

I enjoyed reading your article in the Feb. '02 issue, "Reducing Loudspeaker Enclosure Vibrations." I thought it was quite thorough, and I hope that you get a lot of feedback from fellow readers on it.

The article reminded me of an enclosure design of mine that I started many years ago but put on the back burner for one reason or another. It was for a two-way satellite using Dynaudio drivers. Because I wanted the minimum amount of panel resonance, I decided to use three layers of ¾″ MDF panels. The panels would be sized to allow the use of 1.5″ radius quarter-round molding along each outside edge. A 3″ diameter solid wood ball would be cut into eighths, and each eighth would go onto each outer corner.

This seemed like a better way to get rid of diffraction than the ¾″ radius quarter-round molding and plastic wood corners I had used in the past. After assembly, but before the installation of the drivers, I would paint the entire enclosure with Fleckstone, to simulate the look of solid granite.

I originally wanted the layers to be a ¾″ MDF inner layer, a ¼″ Sorbothane middle layer, and a pair of ¾″ MDF outer layers. I changed my mind after reading your article.

Obviously, three layers of ¾″ MDF is a good place to start. However, instead of the ¼″ Sorbothane middle layer, I thought that one or two layers of ⅛″ Isodamp C-1002 thermoplastic would be a better choice. And instead of simply gluing and screwing the two ¾″ MDF outer layers together, I thought about using North Creek soft glue. Perhaps it would even be possible to use the soft glue to adhere all of the layers together.

I just thought that you might enjoy yet one more possibility for your tests. If you decide to give it a try, and the results look good, then by all means put it in a follow-up article. I'm sure I'll enjoy reading that one, too.

Steve Ball
Austin, Tex.

Jim Moriyasu responds:

Thanks for your kind words and feedback. I'm sure your idea would test well, because three layers of ¾″ MDF are pretty effective with yellow carpenter's glue. Cost would stop me from using the Isodamp material, however, and I like the North Creek soft glue as a very viable alternative. Actually, I prefer to laminate two ½″ layers of nine-ply Baltic birch with soft glue in a vacuum press.

Regarding your approach to dealing with diffraction: I'm finishing up a study on cabinet edge diffraction that compares cabinet edges with 1″, 2″, and 4″ radiuses to that of a conventional sharp-edged enclosure. The measurements show there are definite benefits to radiusing the edge of the cabinet with as large a radius as practical.

LONG-TAILED PAIR

Over the past 30 years, I have built and rebuilt low-power amplifiers with a dual triode amplifier/phase inverter driving a pair of pentodes or beam power tubes (usually 6V6GTs or EL84/6BQ5s), and have found the included single-stage long-tailed circuit (*Fig. 1*) to both sound more pleasant and be more stable than the more common paraphase and split-load circuits typically used in these amplifiers. The open-loop gain is 25 to 30, and the negative feedback gain reduction, when used with a pair of 6BQ5 output tubes, is about a factor of five.

I invite others who are interested to experiment with it, compare it to their favorite circuits, or possibly improve upon it. Perhaps replacing R5 and R6 with a 1.5mA transistor current source may be an interesting endeavor. When self (cathode resistor) biasing the output tubes, without providing a means of balancing the idling currents of the tubes, I prefer to give each tube its own cathode resistor and bypass capacitor. This generally results in more nearly equal currents and better balance of the tubes throughout their lives than using a common resistor.

Regarding "FM Stereo Signal Generator" in the May '02 issue (p. 32), a better basis for this project may be the MPX 2000 transmitter featured in *Poptronics* magazine, July 2001, p. 29. This unit has a microprocessor-controlled, crystal-referenced synthesizer for frequency control, and uses an MC1496 balanced modulator, along with other more common ICs, for the stereo generator circuits. As a result, the MPX 2000 will probably give better performance than the Ramsey FM10A. The MPX 2000 is available from North Country Radio, www.northcountry.com.

Sadly, FM radio to me is no longer a source of audiophile quality music. Just over a year ago, Chicago's second-to-the-last remaining classical FM station was sold to a national chain for $165 million and now plays the works of the Rolling Stones, the Electric Light Orchestra, and other artists of that type. The other classical FM station is under the wing of the local PBS TV station and is still thankfully sticking with classical music, but the audio is quite compressed and sounds dull. The other stations across the dial are mostly various rock formats, plus a few jazz, country, religious, and ethnic programs. Some of them have tolerable sound quality, but most are not only compressed but actually distorted and mushy-sounding.

I just reread an editorial entitled "The Decline and Fall of FM" (*The Audio Amateur*, 4/77, p. 3), and can only say the situation has become even worse than it was then. I am about to rip the FM tuner out of my audio system and pitch it in the garbage.

Michael Kiley
Crestwood, Ill.

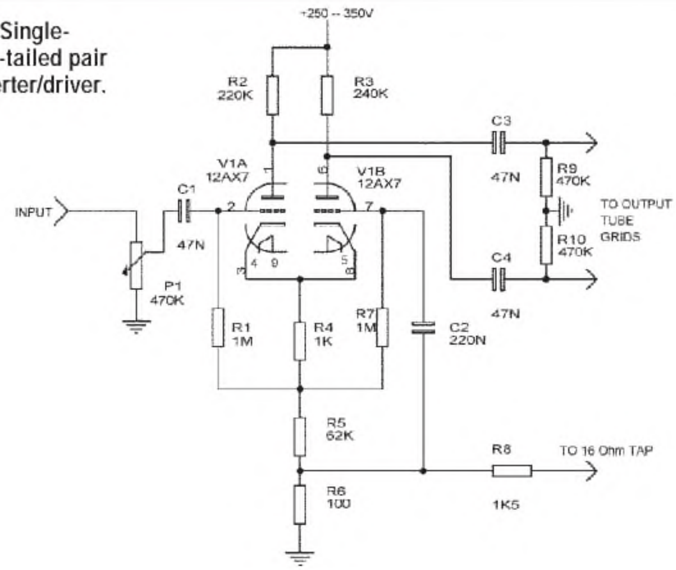
IB DAMPING

First of all, thanks for a very interesting article ("The Infinite Box Concept, Parts 1 and 2," Jan. and Feb. '02). I do not know whether Owens-Corning #705 damping material is available locally but I would like to draw your attention to a material that is readily available almost anywhere and that I have used for front baffle treatment as well as internal box wall lining. This may or may not be suitable for use in an infinite box design such as you have written about.

The material is sold as one side perforated vinyl-coated ¾″ acoustic fiberglass tiles in 2′ × 4′ size. The fiberglass is yellow and very dense. You can use the panels with or without the vinyl film layer, which you can easily peel off. These panels are very inexpensive and available in almost all building-supply stores.

It would seem that what is needed is a study of the absorption coefficient of a number of different materials along

FIGURE 1: Single-stage long-tailed pair phase inverter/driver.



with the useful bandwidth of those materials. Then you could achieve both a desired amount of attenuation and bandwidth via a suitable combination of the desired materials.

I am curious to know whether you have found any correlation between the optimum "dead air space" and any Thiele/Small type box design volume programs? I wonder whether the "dead air space" might be necessary to produce a more uniform pressure front upon the fiberglass damping material. In other words, can you reduce or eliminate the "dead air space" if some form or method of diffusion of the drivers back wave could be devised so that an even pressure front could then be presented to the resistive fiberglass material?

Lastly, I also wonder which drivers you used in your study.

Moray James Campbell
Alberta, Canada

R.O. Wright responds.

I appreciate Mr. Campbell's interest in the infinite box article and the damping material. The acoustic fiberglass tile (AFT) that Mr. Campbell is referring to comes in many formats. Research with the local sound contractors and building-supply outlets (Knoxville, TN) confirmed this fact. The only way I know of to find out whether the fiberglass used on any specific AFT will be useful is to obtain physical data from the manufacturer. If this is unavailable, then you can run a simple experiment to determine the density of the fiberglass.

In order to approximate the basic specifications of Owens-Corning #705, the fiberglass must have the following attributes: (1) density of approximately 5–7 pounds per cubic foot (2) tangled fiber construction (3) a composite thickness, either actual or build-up, as described in our article. If these criteria are met, the material may have the same basic sonic characteristics as Owens-Corning #705. I believe that continuing research in the field of high-density sound damping material is a major factor to superior-sounding speakers.

G.R. Koonce responds.

I want to thank Mr. Campbell for his interest in our infinite box (IB) concept article. Currently, the only dense damping material we have studied is the Owens-Corning (OC) #705. Subject to the density question, the vinyl-faced-fiberglass tiles Mr. Campbell describes may well be fine for building an IB. Some of the OC #705 material I used had come "faced" with vinyl, which stripped off easily. Do watch out that some of the ceiling tiles do not measure the full 2′ by 4′ by 1″ thick of the OC #705 sheets we used.

We are currently gathering a variety of types of fiberglass material that we hope to test. If we gather sufficient data, we will prepare for a follow-up article.

At present, the only advice I can give for working with alternate damping materials is to judge by density. Any fiberglass-based material that has a density of about 6 pounds per cubic foot (P/CF) should perform about the same as the OC #705. Weigh a sheet of your material (stripped of any facing) and then compute what one cubic foot of the material would weigh, and you should have the density.

If you are thinking about the normal fiberglass used for home insulation and guessing it would take a room full to weigh six pounds, that is not the proper material. The high-density materials are rather firm, and we have even heard you can use certain types to construct *the enclosure walls. Materials of density less than 6 P/CF might require use of more than a 4″ thick layer or a modified dead-air volume—we just don't know at this time. Feedback from others building with the IB approach would surely be helpful.*

We studied all sorts of approaches to see how to apply the driver T/S parameters to the design of an IB. The only approach we found that worked for the data we had was the one described in the article. You use the driver T/S parameters to design a lossless (unstuffed) closed-box system of the desired Q_{TC}. The optimum dead-air volume is then about 74% of the design closed-box volume. This was, of course, based on the OC #705 material.

I do not believe the function of the dead-air volume is anything as complex as control of the pressure front, as Mr. Campbell proposes. Placing the OC #705 damping material right at the back of the driver works just fine as a highly damped IB system. The problems are the system response will tend to have a large peak and a rather high f_3. Putting in the dead-air volume helps to limit these problems. I suspect it is more a lumped parameter effect; i.e., introducing an acoustic compliance before the acoustic resistance of the damping material.

Trying to handle the dead-air volume effect theoretically is beyond my abilities. We have heard that using a layer of damping-material densities right behind the driver worked well. Thus, the layers closest to the driver would be low-density going to high density for the rear layers. I did test a layer of very-low-density fiberglass placed ahead of the OC #705 material; it showed little effect. I believe using the volume of dead air is the easier way to build a good IB system.

When we started the IB investigation, we had no idea of what driver characteristics would be required. I thus started testing with a variety of drivers that were on hand. The drivers used are identified in Table 1*, including their purchase date. It is unlikely that any of these drivers are available today.*

Remember that the Ref. #109 and Ref. #139 drivers were used in the constructed IB systems discussed in the article. Since that article, we built three additional IB systems. We built a two-way system using the Ref. #119 (Carboneau #24882) 8″ woofer with a Vifa D27TG-35-06 silk-dome tweeter which worked well.

If you can still obtain this woofer, the dead-air volume was 1062 in^3 and the dimensions of the four 1″ thick damping layers were 10.5″ × 13.25″. The other two systems were subwoofers using drivers from Parts Express. One used the Dayton #295-185 12″ driver and the other the Dayton #295-190 15″ driver. See "The Infinite Box: Constructing a Subwoofer," Part 1, April and May '02 audioXpress.

TABLE 1
IDENTIFICATION OF DRIVERS USED IN IB CONCEPT TESTING

DRIVER	SIZE	SOURCE/MANUFACTURER	PURCHASE DATE:
Ref. #81	6.5″	Pyramid W-61	1989
Ref. #82	6.5″	MCM #55-057	1988
Ref. #109	6.5″	Parts Express #299-125	1994
Ref. #111	6.5″	Parts Express HX-295-220	1996
Ref. #102	8″	Pioneer B20FU20-54F	1996
Ref. #119	8″	Carboneau #24882	1993
Ref. #139	8″	Toutant T816R (Modified)	1982
Ref. #141	8″	Radio Shack #40-1265C	1992
6 × 9	6 × 9″	Advent 6.0i	Old

Thanks for a great article. I've heard some good sound from these types of enclosures, and I've built a couple myself, so this information is first rate. I have a theory on why "From about 150Hz up, the rear leakage shows major dips and other roughness (p. 18)."

I believe these dips are due to interference between the sound from the front of the driver and the rear of the driver. The major dips would occur when the sounds from the front and the rear of the driver are equal in magnitude and with a phase difference of 180°. The rear leakage is measured close to the rear panel, so the sound from the front of the driver should be weaker because it has traveled further and thus filled more volume. But the sound from the rear of the driver has been attenuated by the rear damping (Fig. 31). So it is possible for the sound from the front and rear of the driver to be at the same magnitude near the rear panel.

Near the rear panel, the phase of the sound from the front of the driver will be delayed by the distance that the sound must travel around the enclosure. This is longer than the distance from the rear of the driver to the rear panel. But the sound from the rear of the driver is delayed by the damping material. If the delay through the damping material is the same as the delay due to the additional distance from the front of the driver, then the sound from the front and rear of the driver could differ by 180°.

From Fig. 41, I estimate that the delay through the damping material is about 1.5ms at 150Hz. This would correspond to a distance of about 18″, which is the additional distance that the sound from the front of the driver would need to travel. Most of the sound from the front of the enclosure will not travel by the shortest route to the rear of the enclosure, but will spread out, and thus travel a longer route in getting to the rear of the enclosure. So a distance of 18″ would probably represent an enclosure dimension of about a foot or so. This is close to the enclosure width of 13″.

Figure 41 shows that the delay through the damping material changes with frequency. This would mean that there could be more than one frequency at which there is an interference dip in the rear leakage. Indeed, most of the curves of rear leakage show more than one major dip.

Most of the curves showing major dips use two or more layers of damping material. One layer of damping material has less absorption than two or three, so I would expect that the dips of the rear leakage would be smaller. Figure 23 confirms this.

Finally, the authors state, "The dip frequencies do not match any DA volume resonance mode we could develop" (p. 18). I agree, and I would appreciate any comments they might have on my suggestion of interference causing the major dips in rear leakage. My caveat with an "Infinite Box" is that I would prefer the damping material to absorb more than 30db of rear sound above 200Hz, and my experience tells me that this means a large enclosure.

Dick Crawford
Los Altos, Calif.

G.R. Koonce responds:

I would like to thank Mr. Crawford for his interest in the infinite box (IB) concept article as evidenced by his detailed letter. Mr. Crawford proposes a cause for the previously unexplained major dips that show up in measuring the rear leakage. He concludes they are caused by interference between the actual rear leakage and "crosstalk" from the cone front output leaking to the microphone location.

The location of the major dips did move somewhat in frequency with various IB breadboard (IB-BB) configurations and with the later constructed IB systems. This would fit Mr. Crawford's conclusion. In the IB-BB work the rear-leakage level above 100Hz was in the range of about 15 to 25dB below the cone level. If the cone response crosstalk were nearly this same level, then Mr. Crawford's predicted interference cancellation would likely occur.

The big unknown here is the cone output crosstalk level at the microphone location when measuring the rear leakage. I decided to try measuring this level using a sealed test box of about the same size and construction as the IB-BB. This box actually uses the same front panels to mount the driver as used for the IB-BB and was covered in an early article (SB 1/81, p. 10).

An 8″ woofer was mounted in the box and the rear of the box covered with a 4″ thick layer of Owens-Corning #705 damping material. The external damping layer was added for two reasons. First, to attenuate any leakage through the box rear wall.

Second, when measuring the rear leakage, the microphone was placed very close to the damping layers. Thus any acoustic energy from nearby reflectors would likely be absorbed rather than reflected back to the microphone. The damping layer on the back of the box thus more closely duplicates the normal BB-IB rear-leakage-test conditions. The test box was placed in the same location as used for the IB-BB testing and cone and crosstalk levels measured.

Figure 1 shows the results. Note that the dB scale has been changed 2:1 from that used on plots in the IB concept article, and the crosstalk curve level corrected as though measuring the rear leakage with the one-half area reducer plate installed. The crosstalk level above 100Hz measures in the range 18 to 40dB below the cone level. Thus there are areas of overlap between the levels of the crosstalk and rear leakage. If this test truly represents the IB crosstalk level, then Mr. Crawford's concept of interference causing nulls would seem valid.

The ultimate test to verify Mr. Crawford's concept is not as easy to implement. This would be to mount the IB-BB in a wall so you could fully isolate the cone and rear responses. If the major dips went away it would clearly show they were due to interference. For now, we must be satisfied they are probably caused by interference.

Mr. Crawford points out that in the IB systems he builds he likes to get the rear leakage above 200Hz down at least 30dB. I surely have no problem with this approach. We were trying to show that systems using a lower thickness of damping material would function just fine. Placing the damping layers and opening on the enclosure rear-wall provides attenuation of the rear leakage via reverse diffraction spreading loss (Fig. 2, Jan. '02 aX, p. 9).

As shown in original Fig. 4 (also Jan. '02 aX, p. 9), suppression of rear leakage by 20dB is sufficient to limit the effect on system response to less than 1dB. This is certainly an acceptable value when you consider the effects of grille frames, enclosure edges, and your room.

I thank Mr. Crawford for his input because I believe he has identified the cause of the originally unexplained deep response dips in the rear-leakage measurement. ❖

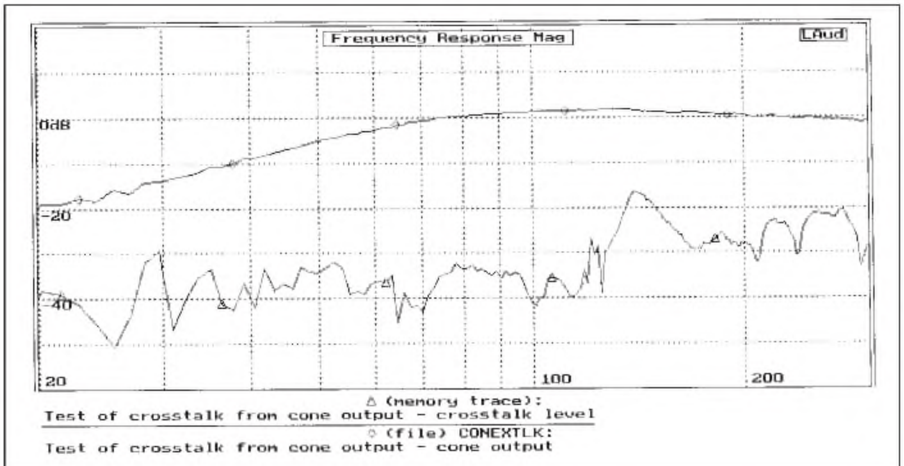


FIGURE 1: Cone output and crosstalk level test results.

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Mike Preamp

from page 17

ings are very clean with a lot of dynamic range.

FINAL THOUGHT

I am very pleased with the results of this project and I think you will find it worth your time and expense to build one. If there are any updates to the design, I will post them on my website for you to download[6].

GLOSSARY

Aliased--When an analog waveform is sampled at less than twice its maximum frequency, the higher frequencies are translated down or aliased into the sampled data.

Dynamic range--The difference in amplitude level between the maximum and minimum useful signals. The maximum level is limited by the overload point of the device under test or the sound card. The minimum level depends on the noise floor.

Intermodulation distortion (IM)--The mixing of two or more pure tones in the device under test to produce sum and difference tones. IM is primarily a test for nonlinear distortion at the highest test frequency.

Noise floor--The residual system noise. Called a noise floor because you can't see any signal with a lesser amplitude.

Total harmonic distortion (THD)--Nonlinearity in the device under test produces harmonics of the test signal. That is, a 1kHz tone would have harmonics at 2kHz, 3kHz, and so on. THD is the square root of the sum of the squares of all measurable harmonic voltage levels. THD can be expressed in dB with respect to the fundamental level or as a percentage of the fundamental level. ❖

Bass and the Room

from page 8

and phase response—this criticism culled from textbooks. However, in practice modern system designers frequently exploit de-tuned, over-damped bass reflex alignments for target functions which are well matched to room gain characteristics and provide a correct overall perceived Q, and which sound subjectively as quick and agile as you could wish for.

With a more complete approach to low-frequency design—this now almost taken for granted by many professionals—the engineering merits of the reflex enclosure may be realized while maximizing sound quality for a given expenditure.

Regardless of the enclosure design adopted, recognition of the room matching aspect is now essential when designing for a given low-frequency system alignment. Low-frequency alignments do matter, but their design should be flexibly interpreted as a part of the overall acoustic system, invoking the target function approach. ❖

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Back issues of *Audio* magazine in excellent condition. These have been kept in my home library and used rarely for either reference material on the various construction projects (authors such as Pass, Jung, and Leach often published in *Audio* first), or to check the Annual Equipment Directories for details, specs, and pricing on used equipment. Complete sets for the years 1974–1994 plus Dec. '68, Feb., Mar. '70, and July–Dec. '73. 260 magazines total. $100. If interested, please e-mail me: gort@softcom.net.

AD INDEX

Audio equipment: Luxman C135 MkIII tube preamp, M1500 SS 75W/ch power amp w/meters, T130 AM/FM tuner; Acoustic Research A-04 integrated amplifier; Revox A-77 Mk IV R-R ½ track high speed (7.5/15 IPS); Nakamichi 500 2-hd cassette recorder; Quad 50E 50W monoblock lab amplifier, transformer coupled, 2–200Ω; Studer D424-2 MO disk recorder 32/44.1/48kHz 16/18/20/24 bit stereo. Test equipment: Bell Model 600 Hall-effect gaussmeter, no probe; Bruel&Kjaer 2305 graphic level recorder and 50dB pot, 1607 frequency and distortion bridge, 1022 beat frequency oscillator, 1024 beat frequency oscillator, 2107 frequency analyzer, 2112 frequency analyzer; General Radio 1340 pulse generator, 1808 precision AC millivoltmeter, 1309B audio oscillator, 1390 random noise generator, 1232 tuned amplifier and null detector, 1313 wide-range audio oscillator, 1310A audio oscillator, 1396A tone burst generator; Hewlett-Packard 8062A precision impulse sound level meter, 5532 counter; Midiman 465 amplifier, 5244L counter, SMPTE Time Window; Tektronix 620A X-Y display. Dick Pierce, Dpierce@world.std.com, 781-826-4953.

Gold Ribbon Concepts drivers, four total. Late production, never used, new, in original factory packaging. Exotic driver is a push-pull 30" line source with gold conductor vacuum evaporated onto Kapton diaphragm. $1500 obo for all 4. Also, new in box Dynaudio 24W100 woofers, four for $400 obo. bitin@attbi.com.

Beveridge System 3 electrostatic/dynamic pair. Dyna Mk III pair. Dyna Stereo 70. Dyna Mk IV. Spare tubes. Pyramid PA-600X stereo basic. Pyramid SEA-3000 3-way stereo crossover. Citation I control center. Fair offers considered. NE Ohio. Prefer pickup/area delivery. For details, e-mail rcmkem@raex.com.

Printed circuit boards for your tube audio projects. Several different boards for most Dynaco products. NEW-PCB for the Marantz 8B DIY. e-mail roymottram@aol.com.

Parts collection (approximate): 40 electrolytic capacitors, 80 disc capacitors, 30 high power (10–12W) fixed and variable power resistors, 18 full-size potentiometers, 15 small/medium transformers (half wall warts), $25. Rotary autotransformer, 8A, 0–130V RMS, $22. Two matching HiComp Auto stereo amps, 18W RMS, $15. nad512@erols.com or Buzz at 301-872-5982.

Dynaco FM-5 tuner, $97; Marantz 23 tuner, $129; GE Hi-Fi VCR, $39; HK430 receiver, $99; Philips CDC-875 CD changer with S-1 DAC, $199; Isodrive, $29; Simply Physics Byteline III digital interconnect, $49; CD-4 laser, $29; TEAC A-106 cassette, $39. Dave 724-274-8149.

WANTED

Single *Glass Audio* issue: Volume 10, Number 1, 1998, to complete my collection. Will trade from my duplicate *GA*s—30 various issues from 1989–1998. E-mail: JAGUGL4546@aol.com, or call (845) 947-2748. John Agugliaro.

Bogen Challenger CH30 mono amp and/or schematics. Have one, need another for stereo. scarinci@juno.com.

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Classic Circuitry
Grommes Hi-Fi

Circuit diagram of the "classic" Grommes Stereo Precision Amplifier, Model 209. The developer, Precision Electronics, started in 1946 and produced hi-fi until 1972, and now manufactures only public address amplifiers.

Courtesy of Tom Tutay, Ft. Walton Beach, FL 32549, and A.A. Hart, Chief Engineer, Precision Electronics, Franklin Park, IL.