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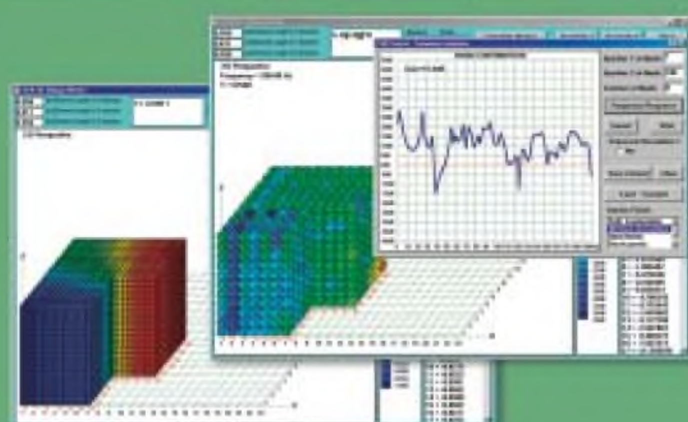

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Behind the Scene

Dr. Joseph D'Appolito has been working as consultant for Usher Audio since early 2001. 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 tests 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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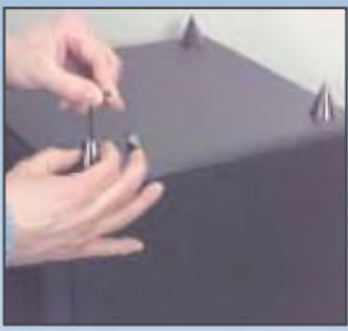

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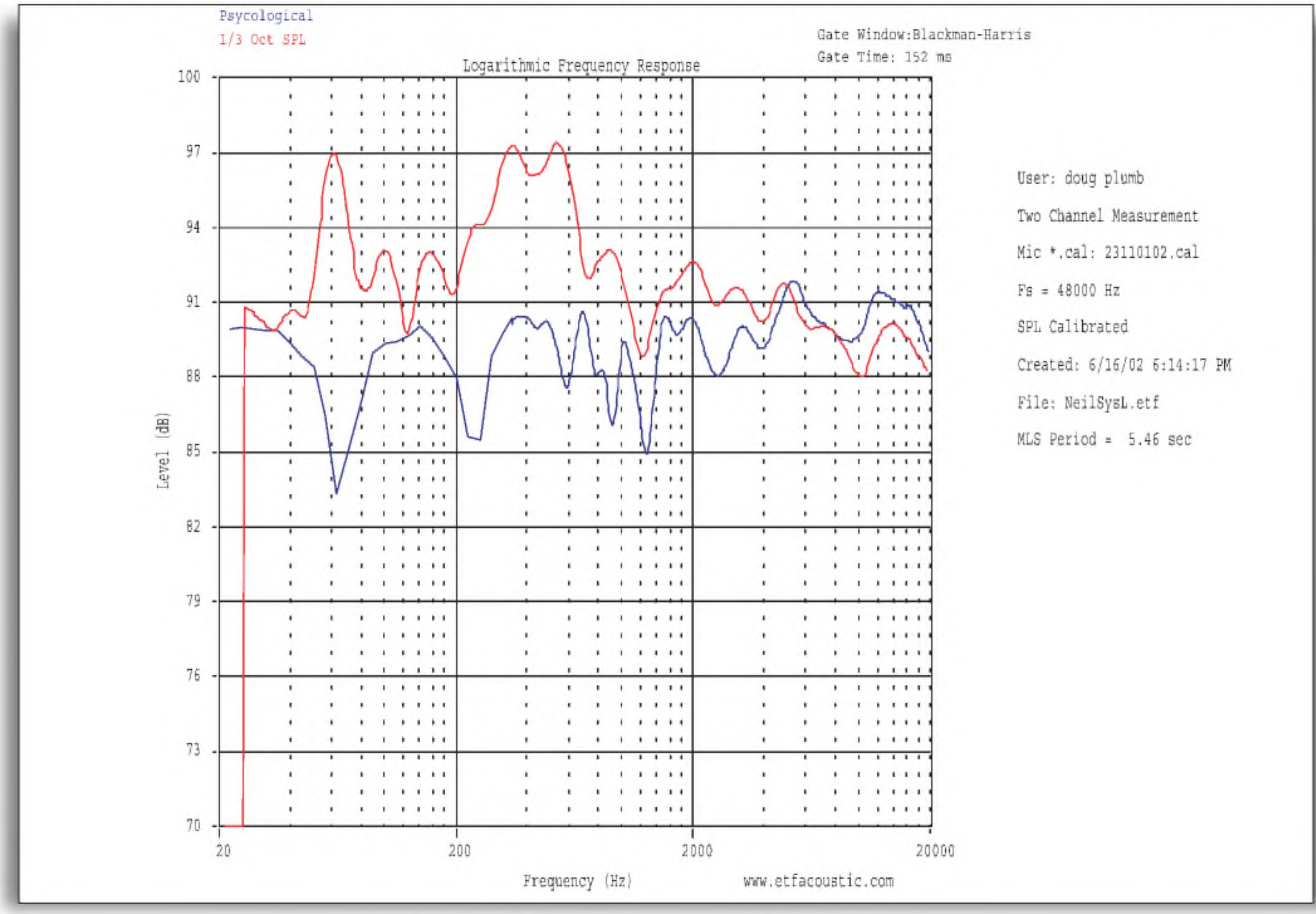

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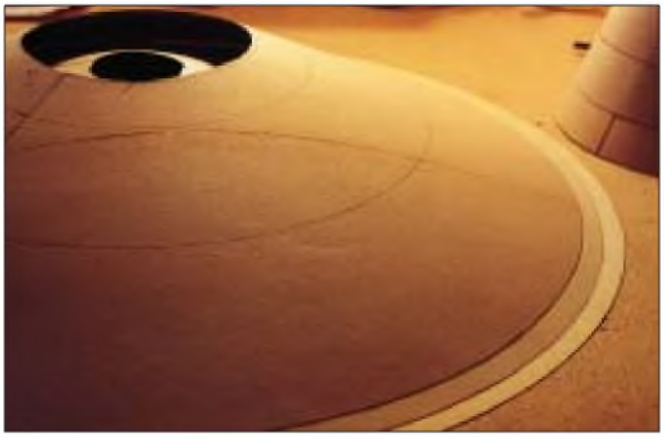

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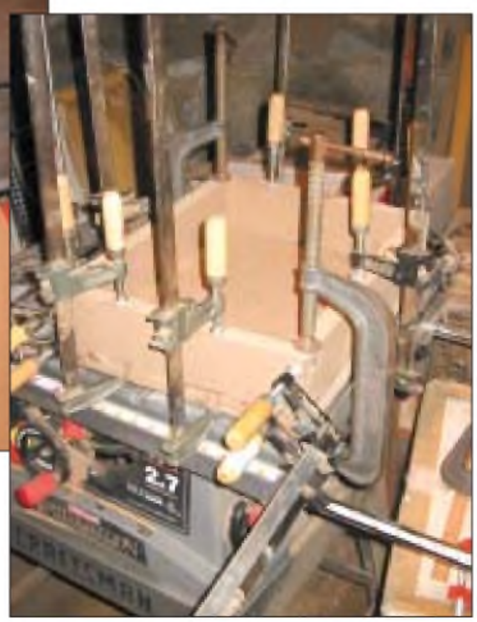

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Regular Contributors

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Amy Wyman	Production Assistant
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Advertising Department
Strategic Media Marketing
1187 Washington St.
Gloucester, MA 01930
Peter Wostrel
Phone: 978-281-7708
Fax: 978-283-4372
E-mail: peterw1@gis.net
Nancy Vernazzaro
Advertising/Account Coordinator

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JOHN STUART MILL

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DASP-2002 Digital Audio Signal Processor

This circuit can be included in every S/PDIF link between two digital audio devices (CD player, DVD player, CD recorder, or MD recorder). Besides allowing the notorious copybit to be modified, it can be regarded as a sort of universal processor for digital audio signals.

Design by T. Giesberts

Unlike a similar circuit design previously published in *Elektor Electronics*, this new design is thus truly more than just a "copybit killer." Naturally, this application is the key feature here–certainly since we have become aware that the DVD Serial Copy Management System often makes it impossible to make the permitted single copy. For audiophiles who want to compile and process their own versions of digital audio signals, the copybit is thus always a major hindrance.

However, the circuit described here has more to offer. For instance, the audio data can also be processed, possibly using a DSP, and the circuit can also convert a coax signal to an optical signal and vice versa, which is a handy extra feature that makes this device quite a bit more universal in use.

As far as manipulating the digital data is concerned, we can note up front that not only is it possible to modify the copybit, it's also possible to replace nearly the complete content of the channel status register (which includes the copybit). Only the data related to the sampling rate, emphasis, and word length are always passed through unchanged. The user bit and validity bit can also be modified. With the user bit, by the way, the most that can be contemplated is to totally clear the bit. Since both channels are used in the consumer format, the total user-bit data stream amounts to 88,200 bits/s for a CD.

The most important task of this circuit is to modify the channel status bit, which involves separate bits for subframes 1 and 2. However, in practice the same information is placed in both data blocks. Most receivers usually check only one of the two subframes, so the received data for the two subframes is set or modified identically.

The only limitation of this circuit is that the modifications are made per channel status block (192 frames), since the user channel data is not tied to the size of this block. For more information, refer to the consumer format specification IEC 60958-3. The validity bit is also present in each frame and can be modified, if you desire.

DESIGN

We intentionally kept the design of this circuit (*Photo 1*) as universal as possible in order to provide avid experimenters a broad range of options. Readers who always take a quick peek at the full schematic (*Fig. 1*) will have noticed that although the schematic is reasonably extensive, most of it is contained in a limited number of ICs. At the risk of oversimplifying, you could say that the circuit consists of an integrated S/PDIF receiver (IC1) and an integrated S/PDIF transmitter (IC2) with a binary counter (IC6) and an EPROM (IC9) fitted between them (between the channel status outputs CI, UI, and VI of IC1 and the channel status inputs VO, CO, and UO of IC2).

You can regard the EPROM as the actual heart of the circuit. A four-way DIP switch for the address inputs of the EPROM allows you to select 16 different tables. The default settings of the

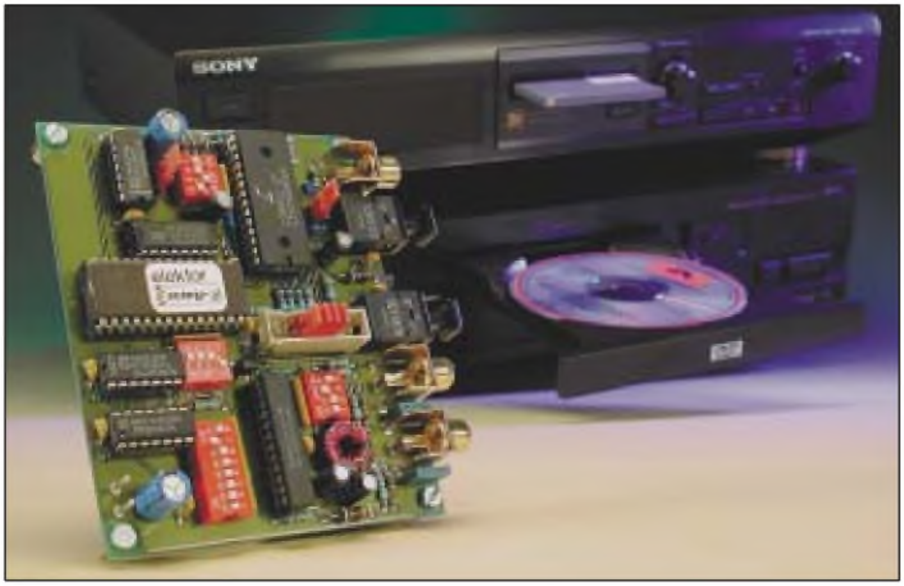
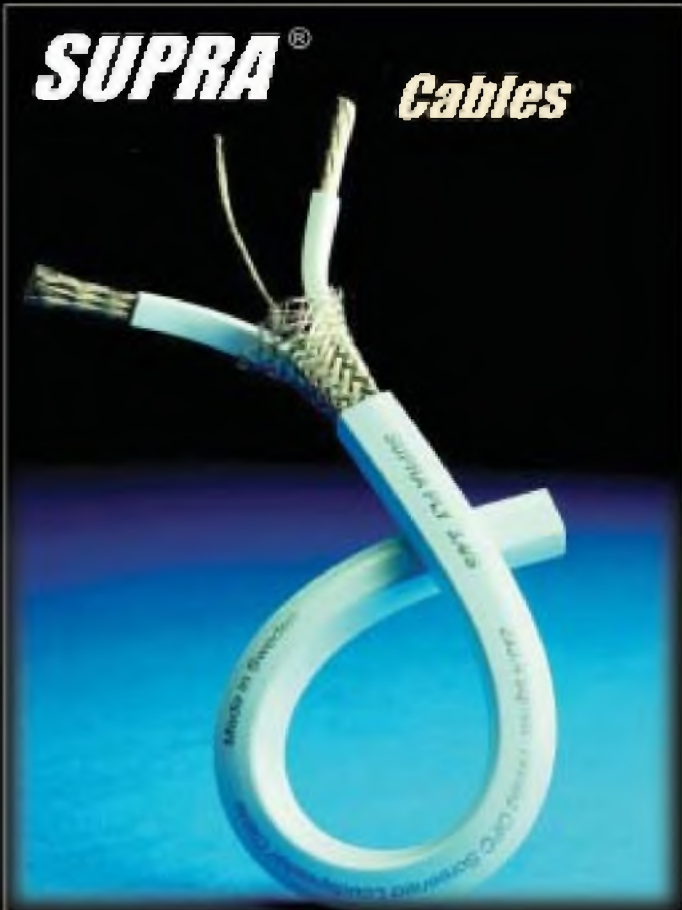


PHOTO 1: A "universal" digital signal processor.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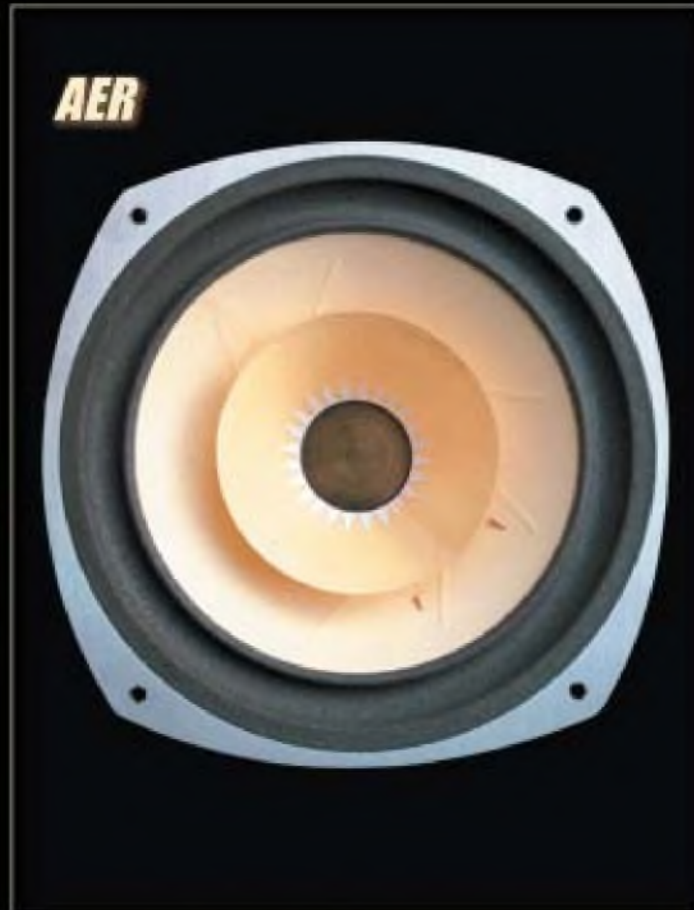
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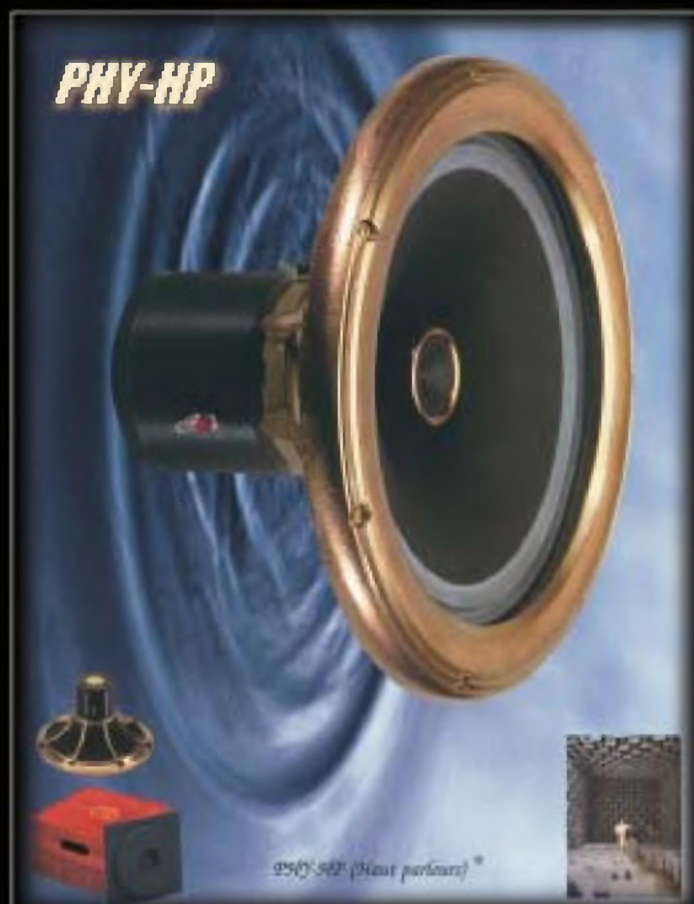
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four DIP switches to enable the circuit to function as a copybit killer are listed in *Table 1*. Currently, only the first three tables are used.

S/PDIF RECEIVER AND TRANSMITTER

For IC1 and IC2 we selected a set of standard Crystal ICs (www.cirrus.com), consisting of a CS8412-CP receiver and a CS8402A-CP transmitter. Thanks to these ICs, the circuit is suitable for use with sampling rates ranging from 32kHz to 48kHz. We have used these two ICs fairly often in past projects, so for more detailed information about them, refer to the manufacturer's data sheets and the articles "Digital VU Meter" (April 1996) and "Sampling Rate Converter" (October 1996). Here we can suffice with a brief description.

The CS8412 is a CMOS IC specially designed for receiving and decoding audio data in the usual standard formats (AES/EBU, IEC958, S/PDIF, and EIAJ CP340). This IC, which has been given the handsome name "digital audio interface receiver" by its maker, receives the signal via an RS422 serial interface and derives the clock and synchronization signals from the received signal. There is support for 256× oversampling, since the clock frequency of the output signal is 256 times the frequency used for sampling.

The audio and digital information is also demultiplexed in an effective manner. The CS8412 demultiplexes the channel, user, and validity data directly to the serial outputs, with dedicated pins for the most important channel status bits (C, U, and VERF). The audio data is output via a special serial audio port that supports 14 formats. The data is available on the SCK, FSYNC, and SDATA pins, which can be configured using the four control pins M0–M3. As you can see in the schematic diagram, in our case a four-way DIP switch (S1) is connected here for this purpose. *Figure 2* shows a simplified block diagram of the CS8412.

Next we have the CS8402A, a "digital audio interface transmitter." As you have doubtless already guessed, this IC is specially intended to be used to encode and transmit audio data in accordance with the previously mentioned interface standards. It allows the values of the most important channel status

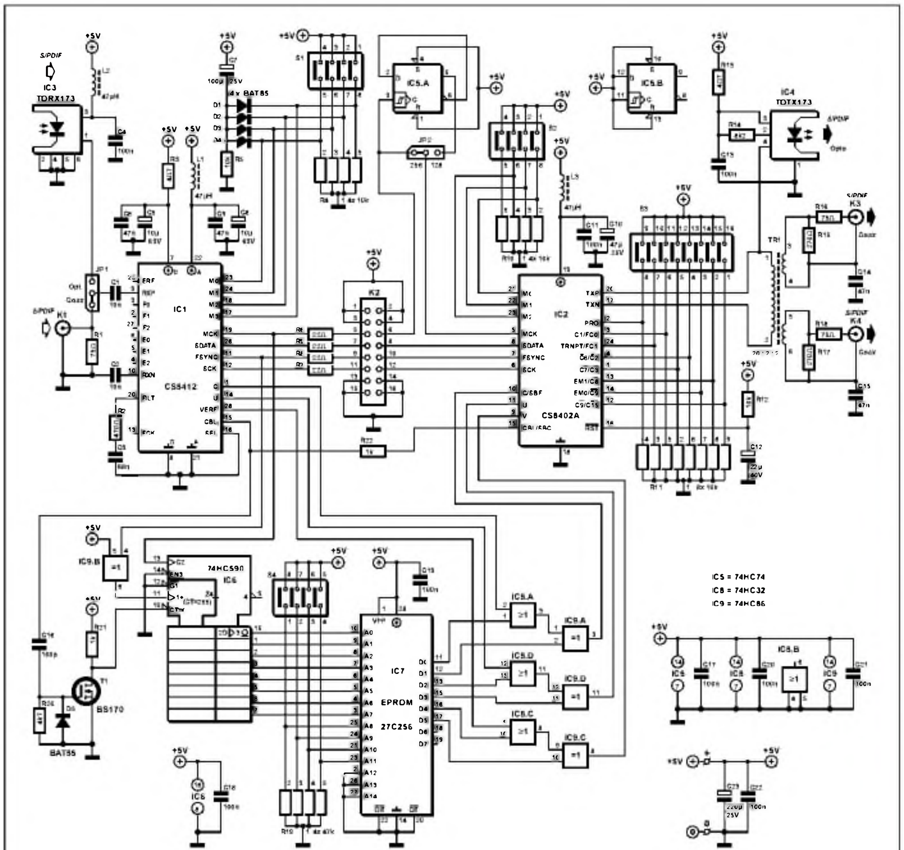

FIGURE 1: The circuit is designed to be as universal as possible. Practically every possible modification can be realized using the four DIP switches.

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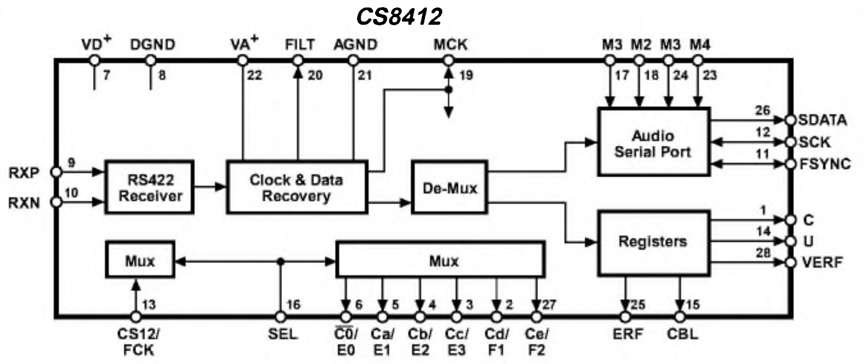

FIGURE 2: Block diagram of the S/PDIF receiver CS8412.

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TABLE 1
THE DEFAULT SETTINGS OF ALL THE SWITCHES AND JUMPERS

S1:	S1-1 OFF	
	S1-2 OFF	
	S1-3 ON	
	S1-4 OFF	
S2:	S2-1 OFF	
	S2-2 OFF	
	S2-3 ON	
	S2-4 N.C.	
S3:	S3-1 OFF	
	S3-2 OFF	
	S3-3 OFF	
	S3-4 OFF	
	S3-5 OFF	
	S3-6 OFF	
	S3-7 ON	
	S3-8 OFF	
S4:	S4-1 ON	
	S4-2 OFF	
	S4-3 OFF	
	S4-4 OFF	
JP1:	coax	
JP2:	256	
K2:	5-6	jumper
	7-8	jumper
	9-10	jumper
	11-12	jumper

bits to be set using seven inputs. You can see these at the lower left of the internal block diagram of the IC (*Fig. 3*).

If you look again at the full schematic diagram in *Fig. 1*, you will see that they can be set using an eight-way DIP switch (S3). All seven inputs also have dual functions, depending on the state of pin 2 (PRO). This input (which can be switched using S3-8) determines whether the IC operates in the "professional" or "consumer" mode. In both cases, the audio data is encoded using the associated applicable standard. In the professional mode (S3-8 open), a

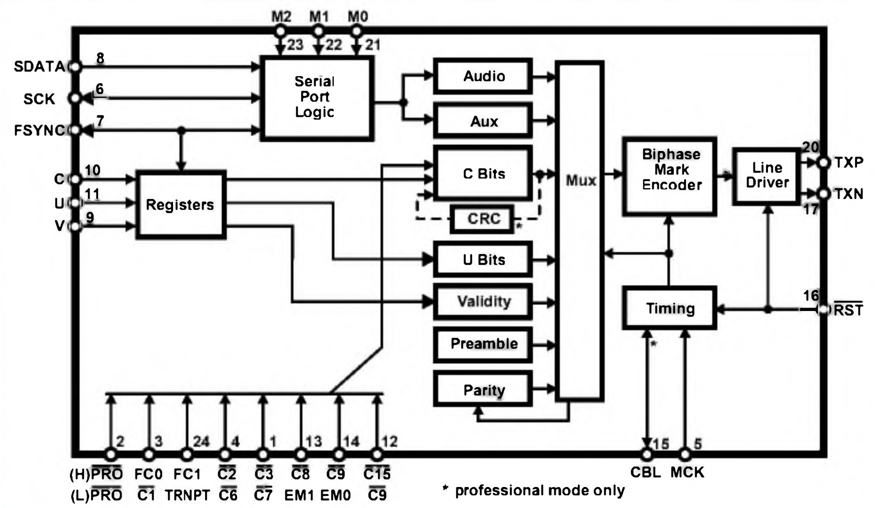

FIGURE 3: Block diagram of the internal structure of the CS8402A.

TABLE 2
THE CONTENT OF THE CHANNEL STATUS BLOCK FOR THE CONSUMER FORMAT

Byte	Consumer format channel status fields							
0	Pro/con = 0	Non-audio = 0	Copyright	Emphasis			Channel status mode = 00	
	bit 0	1	2	3	4	5	6	7
1	Category code							
	bit 8	9	10	11	12	13	14	15
2	Source number				Channel number			
	bit 16	17	18	19	20	21	22	23
3	Sampling frequency				Clock accuracy			
	bit 24	25	26	27	28	29	30	31
4	Word length				(Future original sampling frequency?)			
	bit 32	33	34	35	36	37	38	39
5–23	Reserved							
	bits 40–191							

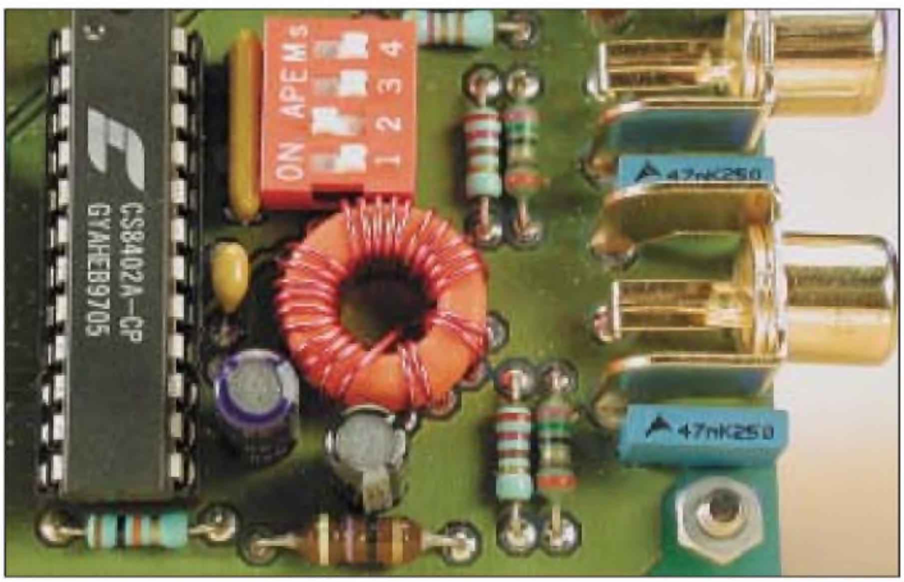

PHOTO 2: Transformer Tr1 can easily be wound on a small toroidal core.

COMPONENTS LIST

RESISTORS:
R1, R16, R18 = 75Ω
R2 = 470Ω
R3, R13 = 4Ω7
R4, R10 = 4-way 10kΩ SIL array
R5, R12 = 10kΩ
R6–R9 = 22Ω
R11 = 8-way 10kΩ SIL array
R14 = 8kΩ2
R15, R17 = 270Ω
R19 = 4-way 47kΩ SIL array
R20 = 4kΩ7
R21, R22 = 1kΩ

CAPACITORS:
C1, C2 = 10nF ceramic, lead pitch 5mm
C3 = 68nF
C4 = 10000nF ceramic, lead pitch 5mm
C11, C13, C17–C22 = 100nF ceramic
C5, C8 = 10µF 63V radial
C6, C9 = 47nF ceramic, lead pitch 5mm
C7 = 100µF 25V radial
C10 = 47µF 25V radial
C12 = 22µF 40V radial
C14, C15 = 47nF
C16 = 100pF
C23 = 220µF 25V radial

INDUCTORS:
L1, L2, L3 = 47µH

SEMICONDUCTORS:
D1–D5 = BAT85
T1 = BS170
IC1 = CS8412-CP Crystal/Cirrus Logic (RS Components)
IC2 = CS8402A-CP Crystal/Cirrus Logic (RS Components)
IC3 = TORX173 (Toshiba)
IC4 = TOTX173 (Toshiba)
IC5 = 74HC74
IC6 = 74HC590
IC7 = EPROM 27C256, programmed, order code 020091-21
IC8 = 74HC32
IC9 = 74HC86

MISCELLANEOUS:
JP1, JP2 = 3-way pinheader with jumper
K1, K3, K4 = cinch (RCA) socket, PCB mount (e.g., T-709G from Monarch/Monacor)
K2 = 16-way boxheader with four jumpers
S1, S2, S4 = 4-way DIP-switch
S3 = 8-way DIP-switch
Tr1 = primary 20 turns 0.5mm ECW, secondary 2 × 2 turns on Ferroxcube core TN13/7,5/5-3E25

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CRC code can also be generated (channel status bit 23), as indicated by the dashed outline in the block diagram.

The serial input (pins 6, 7, and 8) can process seven different formats and audio samples with lengths of 16 to 24 bits. The format can be specified using M0, M1, and M2, and in *Fig. 1* you can see that here again we use a DIP switch (S2) for this purpose. The serial inputs for channel status (C), user data (U), and validity (V) are connected to the EPROM (IC7) via several gates (IC8 and IC9)—we'll say more about this shortly.

Figure 1 also shows that the symmetrical TXP/TXN output of the CS8402A is converted into two standard S/PDIF outputs K3 and K4 (0.5V pp/75Ω) using a (homemade) transformer Tr1. The electrical isolation provided by the transformer has the additional advantage of preventing earth loops. An optical output is also generated using IC4.

JUGGLING BITS

Using jumper JP1, you can select whether the coax input signal or the optical input signal is to be fed to the receiver. Beside the audio data, IC1 also decodes the U, C, and V bits. The VERF output represents the received V bit OR'ed with the internal error flag ERF, which is active in the event of a parity error, biphase coding violation, or unlocked PLL receiver. An active V bit means that there is something wrong with the associated audio sample, in which case the data can be muted by the DAC or recorder.

These three bits are connected to a number of gates (IC8/IC9), by means of which four states can be defined for each bit. For each frame (using the same option for each subframe), each bit can be sent to the transmitter unchanged, inverted, as a fixed "0," or as a fixed "1." An OR gate, an XOR gate, and two data bits from the EPROM are used to generate each of these four combinations: D0/D1 for the channel status bit, D2/D3 for the user channel bit, and D4/D5 for the validity bit.

As an example, let's look at the combinations of D0/D1 for the C bit. With the combination 0/0 (no change), CI is passed through unchanged by the OR gate (IC8a) and the XOR gate (IC9a). With 1/0 (constant "1"), there is a "1" at the outputs of both gates. With 0/1 (in-

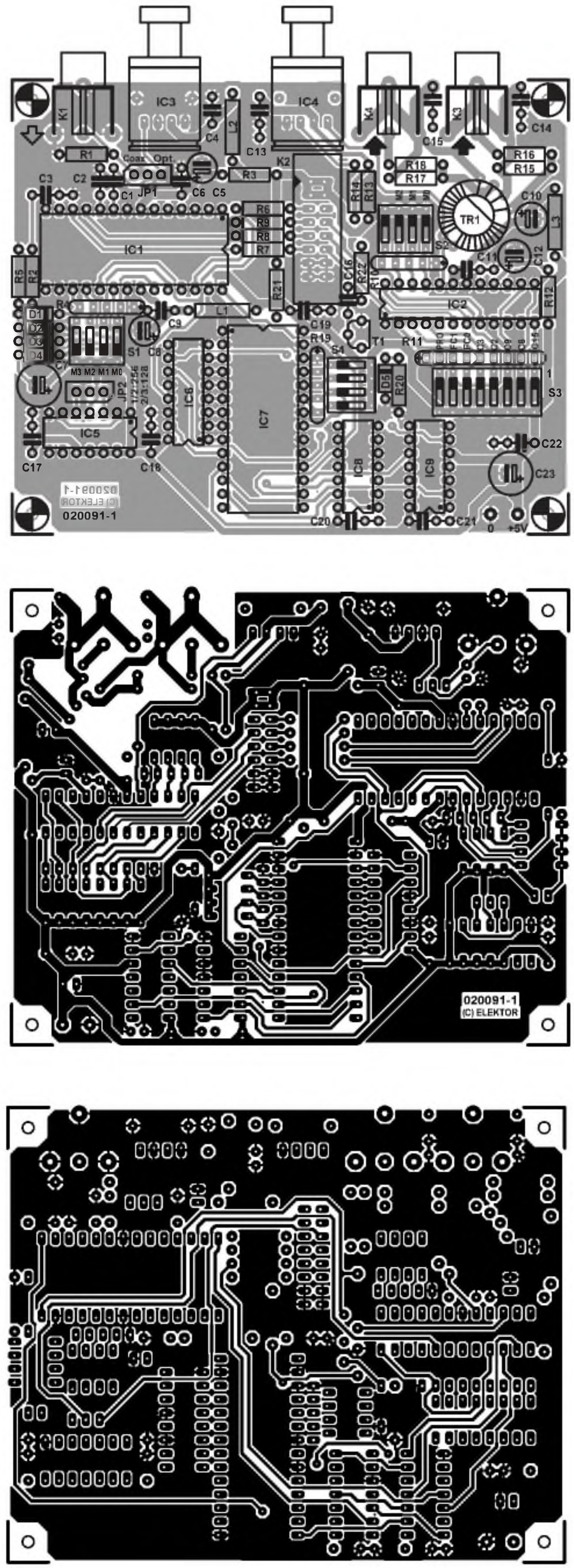
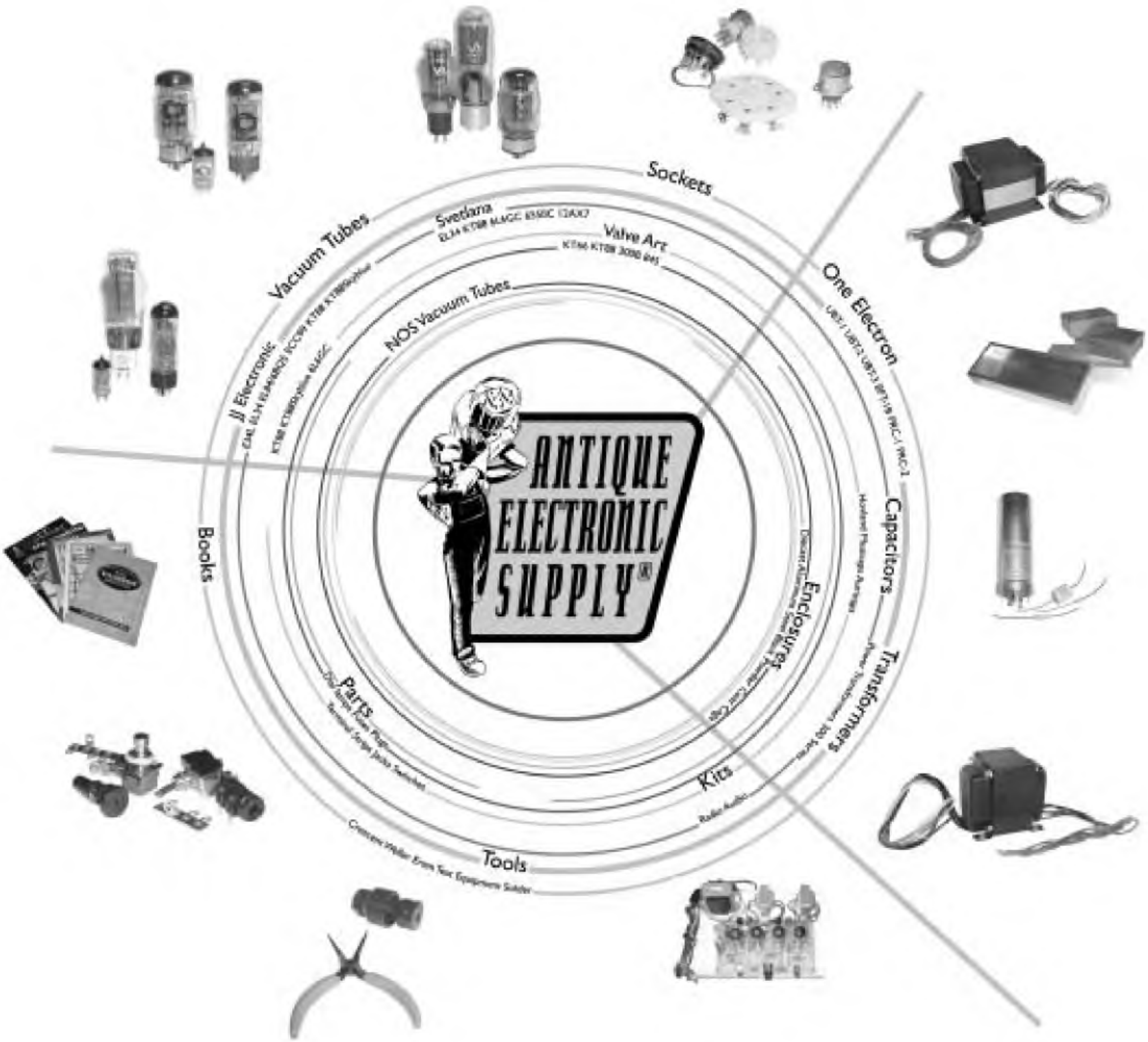


FIGURE 4: The double-sided printed circuit board is relatively compact and easy to assemble.

A-2153-4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verted), CI is passed through by the OR gate but inverted by the XOR gate. Finally, with 1/1 (constant "0"), there is a "1" at the output of the OR gate, which is inverted by the XOR gate.

The outputs of the gates are connected to the transmitter inputs having the same designations. In order to be able to use these bits, the transmitter must be set to the professional mode (S3-8 open = OFF) and also be switched into transparent mode (S3-7 closed = ON). The rest of S3 then no longer has any function.

An eight-bit binary counter with an output register (IC65, a 74HC590) is used to address the EPROM. The counter is clocked by the inverted frame synchronization signal FSYNC. Since the output register is clocked by the master clock MCK, the address for the next frame becomes active only after the transmitter has already latched the second (right-channel) subframe (the delays of the XOR, counter, and EPROM also contribute to the timing).

The "channel status block start" output (CBL) is used to synchronize the counter with the channel status block. A differentiator network composed of R20/C16, T1, and R21 generates a reset pulse that is applied to the counter clear input of the counter to cause it to resume counting from zero (after 192 frames). Since the transmitter CBL pin can also sometimes be an output, depending on the mode, R22 is inserted in this lead to prevent any problems from occurring.

EPROM DATA

The most important aspect of the circuit is determining the data for the EPROM. For this you need to know the composition of the channel status block. *Table 2* shows the content of this block for the consumer mode, with brief explanations of the various bits in *Table 3*. Finally, *Table 4* shows what the channel status block looks like after having been processed by the circuit in the default configuration (the second table in the EPROM).

In the last of these tables, "INP" means the original bit in the input signal (INPUT). "DAT" has been chosen as the category code, since this gave the fewest problems in past experience. The rest is self-explanatory.

Translating the individual bits in *Table 4* into the bit combinations for the EPROM (in combination with the choice of U and V bits) yields the data for the EPROM shown in *Table 5* (msb/lsb: U ⇒ 00, 1 ⇒ 01, 0 ⇒ 11). *Table 5* is incorporated in the EPROM as the second table (S4-1 ON). The first table (S4-1/2/3/4 OFF) allows all data to pass unchanged, for which all data bits in the EPROM are "0."

TABLE 3
SUMMARY EXPLANATION OF THE BITS IN *TABLE 2*

Consumer format channel status field interpretations

Bits	label	Interpretation
0	pro/con	0: consumer; 1: professional format
1	non-audio	0: suitable for conversion to analog audio using linear PCM 1: not suitable
2	copyright	0: asserted; 1: not asserted
3–5	emphasis	000: Emphasis not indicated 100: emphasis—CD-type
6–7	channel status mode	00: mode zero; other values reserved
8–15	category code	The category code depends on the equipment type. For most codes the MSB of the category code carries information about generation status of the material (refer to SCMS), and is called the L-bit. The following are common codes: LSB MSB CD player 1000 0000 DAT player 1100 000L DCC player 1100 001L Mini disc 1001 001L
16–19	source number	(bit 16 is LSB)
20–23	channel number	(bit 20 is LSB)
24–27	sampling frequency	0000: 44.1 kHz 0100: 48 kHz 1100: 32 kHz
28–29	clock accuracy	10: Level I, ±50 ppm 00: Level II, ±1000 ppm 01: Level III, variable pitch shifted
30–31	reserved	
32	word length (field size)	0: Maximum length 20 bits 1: Maximum length 24 bits
33–35	word length	if bit 32 = 1 / if bit 32 = 0 000: not indicated / not indicated 101: 24 bits / 20 bits 001: 23 bits / 19 bits 010: 22 bits / 18 bits 011: 21 bits / 17 bits 100: 20 bits / 16 bits
36–39	reserved	
40–191	reserved	

TABLE 4
THE CHANNEL STATUS BLOCK AFTER PROCESSING BY THE CIRCUIT (USING THE DEFAULT SETTINGS)

	LSB							MSB
Byte 0	0	0	1	INP	INP	INP	0	0
Byte 1	1	1	0	0	0	0	0	0
Byte 2	0	0	0	0	0	0	0	0
Byte 3	INP	INP	INP	INP	0	0	0	0
Byte 4	INP	INP	INP	INP	0	0	0	0
Bytes 5–23	0	0	0	0	0	0		

As a small extra, there is also a third EPROM table (S4-1/3/4 OFF, S4-2 ON) that makes the validity bit always "1". However, we have not tested any recorder that refused the data on this account (the data is otherwise the same as with the second EPROM table). If you have the necessary tools, you can experiment using the remaining 13 tables of the EPROM.

EXTRA FEATURE

As you can see from *Fig. 1*, the audio data passes from the receiver to the transmitter via box header K2. This means that it is necessary to fit four jumpers to this header to loop through four signals (see also *Table 1*).

The presence of K2 also makes it possible to process the audio data if desired—for example, using a DSP. In that case, the data format is naturally determined by the settings of the receiver (default: format 2, I^2S-compatible).

SOLDERING

Well, it's time to pick up the soldering iron. In order to avoid having to insert a lot of wire bridges (one of which is always overlooked, guaranteeing that the circuit won't work), a double-sided printed circuit board has been developed for this digital audio processor. *Figure 4* shows the component layout and the track layouts for both sides. As you can see, the design of the circuit board is well organized and compact. Where there is sufficient room, the functions of the DIP switches are indicated in addition to their numbers.

Be careful with S1, since here S1-4 corresponds to M0. The cinch sockets and optical connectors are all located along one side of the board, which simplifies fitting the board into an enclosure.

Assembling the circuit board should hardly present any problems, and there is actually no reason why you cannot complete it in a couple of hours. There are not any SMD components or other "troublesome" components on the board; you can simply insert the ICs into sockets, and there is no need to fit components on the bottom side of the board.

The only thing that might be considered somewhat difficult is winding the transformer for Tr1. Actually, it's dead easy. For this we use a Ferroxcube (formerly Philips) core, type TN13/7,5/5-3E25.

The primary winding consists of 20 turns, and the two secondary windings consist of two turns each, all using 0.5mm (SWG24 or SWG26) enameled copper wire. Space the 20 turns of the primary winding neatly around the toroidal core, leaving a bit of space in the middle for the two secondaries, which are subsequently wound in between. *Photo 2* gives a good picture of what is intended.

Once you have fully assembled the circuit board and carefully checked it against the component layout drawing and components list, you can connect a suitable 5V power supply to the two pins marked "0" and "+5V." Any power supply that can deliver a reasonably well-regulated voltage at a current of at least 100mA will be satisfactory.

Once you've connected the power supply, it's a good idea to use a multimeter to check whether the desired 5V voltage is in fact present across C5 and C10. If this test is successful, there is a good chance that the rest of the assembly is also in order. Should problems nevertheless arise in use, there must be an assembly error, and this can only be found by careful inspection of the entire circuit board.

Unfortunately, no measurement points or test voltages can be indicated here, since this is a fully digital circuit with the exception of the supply lines. However, if the board has been assembled with due care, it is highly likely that it will work straightway. ❖

PCB layout and EPROM contents available from Free Downloads section at www.elektor-electronics.co.uk. PC board and EPROM, programmed, available from Old Colony Sound Lab (PO Box 876, Peterborough, NH 03458, 603-924-9464, Fax 603-924-9467, E-mail custserv@audioXpress.com) for $43.

TABLE 5
THE TRANSLATION OF *TABLE 4* INTO BIT COMBINATIONS FOR THE EPROM

Byte 0	00000011	00000011	00000001	00000000	00000000	00000000	00000011	00000011
Byte 1	00000001	00000001	00000011	00000011	00000011	00000011	00000011	00000011
Byte 2	00000011	00000011	00000011	00000011	00000011	00000011	00000011	00000011
Byte 3	00000000	00000000	00000000	00000000	00000011	00000011	00000011	00000011
Byte 4	00000000	00000000	00000000	00000000	00000011	00000011	00000011	00000011
Bytes 5–23	00000011	00000011	00010011	00000011	00000011	00000011	00000011	00000011

(All C bits starting with the most significant nibble of byte 4 are set to "0.")

The circuit described here is exclusively intended to be used for the digital recording or processing of personal musical works. The editorial staff dissociates themselves from any unlawful use of this circuit involving the infringement of copyrights resting on audio media such as CDs, MDs, or DVDs.

Modern copy protection systems, such as Cactus Data Shield, SafeAudio, and Key2Audio, generally exploit the characteristics of the drive and thus cannot be affected by the circuit described here.

Building the THORs

Our fearless leader takes us—step-by-step—through the successful construction of the revolutionary THOR TL speakers. **By Edward T. Dell, Jr.**

The completed unit at home.

Building a pair of Joe D'Appolito's THOR transmission line speakers (*aX*, May 2002, pp. 8–27) is not a trivial undertaking. The project may be done more simply than I did it, but these boxes require careful work and attention to detail. The design is not really complicated, but the final result requires that the dimensions be very nearly correct and that the stuffing follow D'Appolito's exact instructions.

The cost of the drivers and crossover components for the THOR can be had in kit form from Madisound and several other vendors for a bit more than $915, not including shipping. Other materials, not including tools, may be had for about $100. I have included a list of materials in *Table 1*, and a list of tools I found useful in *Table 2*.

One of the more difficult tasks in this project is getting the two sheets of MDF into your workshop. It will require two sets of muscles to accomplish this—even if you have the 1″ MDF cut into two pieces. A 38″ × 48″ piece is easier to handle, but the remainder is still pretty heavy for one person. I had my lumberyard make one cut in the ¾″ MDF sheet to 51″, from which I cut the sides.

CUTTING PARTS

A table saw is almost a necessity for producing accurate cuts of the MDF pieces of the two cabinets. With great care, it might be possible to use a straightedge guide with a circular saw to make the cuts shown in Fig. 20, p. 26, in the original article, and reprinted in the July issue, page 67 (*Photo 1*).

The dimensions given are accurate in all cases, with the one exception of the internal baffle—Fig. 1A in the original article, and Fig. 1 in the July reprint. These indicated an 86° angle at the upper end of the baffle where it meets the top of the cabinet (*Photo 2*). I cut mine to that spec, and filled the void between the top toward the front of the enclosure, just before gluing, with good old Plastic Wood®. It worked fine.

The upper cabinet could be assembled using at least three methods: butt joints, screws, or biscuits. Butt joints could be done using a few temporary screws during the clamping and gluing procedure, removing the screws after the joints are dry and filling the screw holes with Plastic Wood.

Stanley makes a pair of simple tools which could work very well in assembling this project. The "Screwmate"® is designed to drill a screw pilot hole as

PHOTO 1: THOR enclosure parts are cut out. From left they include the sides, baffle boards, front panels with angle blocks below, back panels, and tops.

well as a countersink and a hole above the screw which accepts a plug. The company also makes a matching plug cutter which makes neat plugs for the holes from scrap MDF. These pop out easily from the scrap piece with a screwdriver. The only problem is that you really need a drill press to use the plug cutter.

Using 1½ × #8 countersunk screws topped with plugs glued into the holes, you can sand off the excess plug, which becomes virtually invisible on completion. If you use this method, be sure to sink the screws deep enough around the three edges of the front baffle since you will be rounding the edges later with a ⅛″ roundover bit. The same caution applies to where the top is attached to the sides.

PHOTO 2: Closeup of the baffles showing the 86° angle cut.

PHOTO 3: Biscuit cutter and slot. Note the marked center on the tool face.

PHOTO 4: Underside of the top, with overkill on the biscuit slots.

BISCUITS, BISCUITS

Having been impressed by Norm Abramson's work in his "New Yankee Workshop" programs on PBS, I decided to use one of his favorite joint gluing aids: the biscuit. Unfortunately, this method requires very accurate tools, and the cheapest biscuit cutter I've found is just a bit under $100. (There are adapter kits for routers to make biscuit cuts, but look to be more difficult to use accurately and are not useful for angled cuts.) I used #10 biscuits and a DeWalt cutter (*Photo 3*), an excellent tool which works very well. I am a firm believer in buying tools for the long term. I can't remember ever enjoying using a cheap tool–but you may not be so afflicted.

It is a good idea to very clearly mark each piece of each enclosure, i.e., front, back, side, top, bottom, and so on, plus outside and inside, as well as left and right. Mark them in a place that is not going to be cut out for a driver. A heavy carpenter's pencil works well. This will help to ensure that you know where you are in preparation and assembly.

If you decide to use biscuits, as I did, don't overdo it (as I did). They make a very strong joint when used correctly, and two to four to a joint in this project are plenty (*Photo 4*). For the base use no more than two to any joint from sides to the top, and only one on the fronts and sides to each other.

Biscuits require very careful locating marks with a sharp pencil to match where the biscuits will join the two panels. Lay the mating parts side-by-side and mark both at once. Mark the edge-cut locations using a square. Most of these biscuit-cutting tools are set at the factory for a ¾″ vertical center on the most popular thickness of ply and MDF. In making the cuts, use a flat base to make sure edge and face cuts match precisely.

TABLE 1
MATERIALS—ENCLOSURES

4 × 8′ sheet ¾″ medium density fiberboard (MDF)
4 × 8′ sheet 1″ MDF
Small piece 2ft^2 ¼″ hardboard (crossover bases)
1 pt. Franklin Tite Bond® glue
Plastic Wood®
Double-sided carpet tape
1 qt each of primer and finish paint
Assorted sandpaper and tack cloths
16 ¼ × 20 threaded brass inserts and matching ¼ × 20 × 1½″ bolts
32 #8 × ⅛″ Phillips head wood or metal screws, or 32 #8 × ¾″ machine screws with mating T-nuts
100 #10 biscuits (optional)
30 4″ cable ties

CUTS GALORE

I found it prudent to make the first biscuit cut exactly centered on the marked line. The tool has a center marker on its face, front, and top, indicating where the blade middle is located. Then I made cuts just to either side of my marked line to ensure that the biscuits would fit snugly. Be sure to push the tool to its full depth.

Cut the holes for the drivers before assembly. This is a must if you use a router with the Jasper hole cutting jig (*Table 2*). The centering pin must be held in place by a piece of scrap stock under the front panel plus a couple of strips of carpet tape to hold the panel and scrap together. The Jasper tool

TABLE 2
TOOLS

Every builder will have or prefer different tools for this task. My suggestions may be helpful.

1. Table saw. If you already have a hand circular saw, there are guides which might make cutting out uniform panels feasible. A number of inexpensive, direct-drive table saws, Chinese manufacture, are available for under $250, even at Sears in Craftsman grade.
2. Biscuit cutter (plate joiner).
3. Circle cutting guide, Jasper, available from McFeely's (1-800-443-7937), whose catalog is a must for speaker builders.
4. Plunge router and a ¼″ straight bit plus a ⅛″ roundover. DeWalt has a new model with built-in dust collector.
5. Dust collector shop vacuum.
6. Belt and palm sanders, with appropriate sandpaper grits.
7. Electric hand drill and optional 12″ throat drill press, bits.
8. A dozen clamps, four of which should be 5′ pipe clamps.
9. Putty knife.
10. 12″ carpenter's square and tape measure.
11. Jig saw or small hand saw (for tweeter ears cutouts).
12. Screwdrivers.
13. Center punch, mallet, and hammer.
14. Dust mask and safety glasses.

PARTS CONNEXION

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works well, although the guide holes for cutting the smaller openings lie under the router's shoe. Be sure to note that the hole sizes marked on the jig are based on using a $\frac{1}{4}$″ straight bit. This makes good sense, especially when cutting MDF, since the amount of dust is much less with the smaller bit.

The dust problem is serious when working with MDF since the material seems to be made of dust. Be sure to wear a dust mask and safety glasses, and you should also plan to equip your saw with some means of controlling the dust, usually an adapter for the saw and a shop vacuum. If you buy a plunge router, a new one from DeWalt has a dust removal hose connection, which would cut down on the dust quotient in this project in a major way. DeWalt claims this feature is unique. Routers seem to be the only hand tools not equipped with dust collectors, except the hand circular saw.

PHOTO 5: A test cut for the tweeter hole, showing the Jasper rig and the dust mask.

PHOTO 6: The plunge router and Jasper guide do a good job of cutting holes and rim recesses.

DRIVER HOLES

Note that the speaker holes involve two procedures. If you use the router and the Jasper rig to do the rim recesses, you must calculate both the diameter and the depth. Plunge routers have a clamp to set depth of cut (*Photo 6*). It is best to make some test cuts to make sure the depth is accurate.

The peculiar dimensions (like the $6\frac{31}{32}$″ diameter of the mid-woofer outer rim) are a result of translating metric to English measure, and since the Jasper tool is in English measure, with the smallest increments in 16ths, cutting the recesses with it is a real problem. You want the fit to be snug to minimize diffraction. I found that the rims of the larger drivers will fit a diameter of $6\frac{11}{16}$″.

PHOTO 7: Inside the front panel, I rounded driver opening edges, again with too many biscuit cuts on the edges.

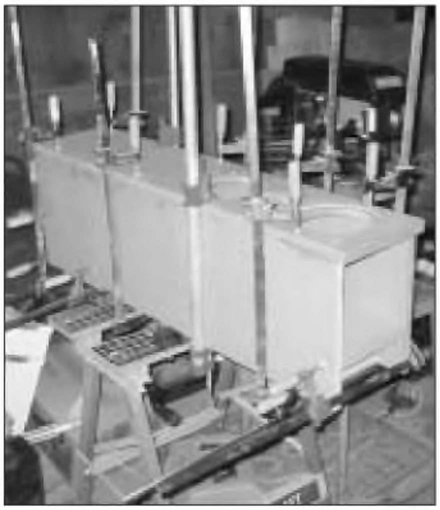


PHOTO 8: There are never enough clamps for a gluing procedure.

Using the Jasper guide to rabbet the recesses requires a larger bit. Just keep in mind that the overall hole diameter marked on the jig assumes use of a $\frac{1}{4}$″ bit–meaning that the circumference at the bit's center is $\frac{1}{8}$″ smaller than the marked diameter. If you choose a larger bit for cutting a wider recess, the circumference of the cut will be larger than the tool's marked dimension by one half the width of the bit, minus $\frac{1}{8}$″. The Jasper manual has instructions for this task, but they seemed fairly difficult to me.

For these reasons I decided to cut the inner holes with the Jasper tool, but to rabbet the recesses. In order to get a final result of the actual size of the mid-woofer rims, which is $6\frac{11}{16}$″, I had to resize the primary opening to $5\frac{11}{16}$″ diameter. The smallest of the ball-bearing collar for the rabbeting tool enables a maximum $\frac{5}{16}$″ wide rabbet, which is right for the mid-woofer drivers, yielding the desired $6\frac{11}{16}$″ diameter rabbeted recess, $\frac{1}{32}$″ deep.

In the case of the tweeter, I found by cutting test holes that the driver would just fit into a $4\frac{11}{16}$″ diameter circular recess, $\frac{1}{4}$″ deep. Therefore, to use a $\frac{5}{16}$″ rabbet, I enlarged the inner hole to $3\frac{5}{16}$″ rather than 3″ indicated in the drawing. This does not affect the mounting shelf at all. As soon as the round tweeter holes are finished, cut out the small recesses for the tweeter "ears."

I also decided to gild the lily somewhat by using a $\frac{3}{8}$″ roundover bit to round both sides of the bottom of the baffle board, as well as the inside of the top of the back panel, and the inside edge of the top panel. This last I did leaving the outer $\frac{3}{4}$″ edges un-rounded. These touches were in aid of smoother transitions for the inevitable diffraction, but I am not about to test the effectiveness of these alterations by building another set without the rounded edges. As David Hafler used to say about adding expensive quality capacitors to his products, "So, a little chicken soup couldn't hurt." I also used the same tool to round the inner edges of the driver openings (*Photo 7*).

A preflight dry assembly of the cabinet parts and the base units is the only prudent path to success. Also, use

clamps in the dry assembly, checking to make sure edges match and there are no gaps. Sometimes a biscuit will need re-seating.

If all is well, you may begin to glue the assembly. I put a dab of glue into each biscuit slot before inserting and adjusting it. The biscuits are compressed wood, which expands slightly when glued.

ASSEMBLY

I joined the back and one side first. I had put biscuits in place to hold one side of the baffle, and installed that next. I then added the second side, carefully positioning the baffle to marks made to locate its second edge. I had also marked the outside for the location of the baffle. I also used glue on this second edge, but supplemented this with sunken screws through the side and into the baffle edge, covered later with plugs glued in place.

The front panel went in place next. I then added the angled plates to the bottom openings with glue and the encouragement of a plastic-faced mallet to position them properly. This was easier to do than I anticipated and my 45° cuts fit almost perfectly. I did not assemble the top at this point, leaving it for later since it is held in place by biscuits attaching to the front and sides only (*Photo 8*).

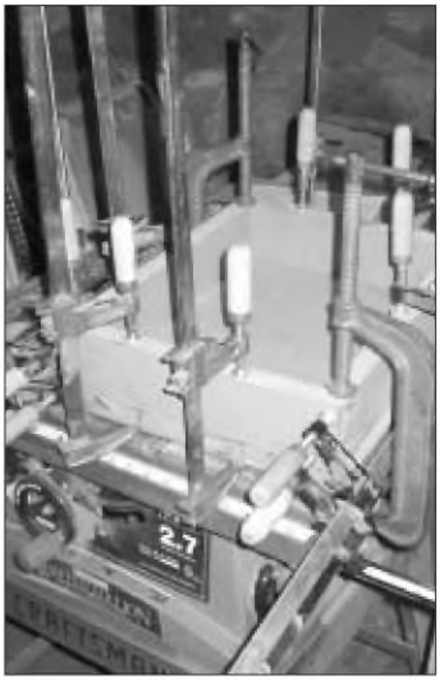


PHOTO 9: Even the simple five-part base requires more clamps than you anticipate.

You work quickly when gluing, although the Franklin Tite-Bond® glue gives you reasonable time to get everything together before the clamps go on. As you clamp be sure to check for squareness of the box by measuring the two diagonals of the bottom, corner to corner, which should be equal. I had 12 clamps for the job and I always think I should have used more whenever I get to the assembly of any speaker box. I let the glue dry for 24 hours.

Before you assemble the bases, cut an appropriate hole in the back panel of the base to accept your input hardware. I mounted pairs of 5-way binding posts on a blank metal wall switch cover. Next, carefully mark the hole locations on the inside of the top for the machine screws, which will attach the base to the cabinet. The dimensions in Fig. 1 of the July issue are accurate. A carpenter's square set to the various dimensions works well.

After center-punching the locations, I drilled the holes on a drill press. The removable base is necessary because of the stuffing procedures. After the cabinet top is added, the bottom of the TL is the only access you will have to install the acrylic material.

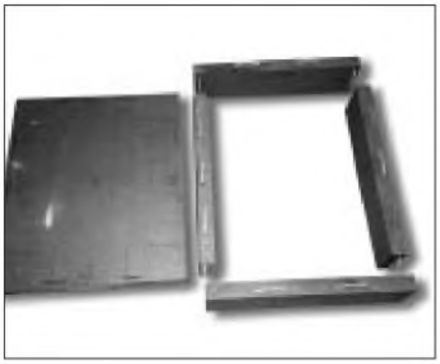


PHOTO 10: Biscuits for the base, overkill city.

PHOTO 11: Bottom side of the tower showing the angled blocks and the brass inserts installed. Note that I used Mortite between the angled blocks and the back and front panels, to seal any void.

MATING HARDWARE

I ordered ¼ × 20 × 1½″ bolts and matching brass threaded inserts to join the base and TL box. This may be overkill, but the finished units are quite heavy

PHOTO 12: Crossovers in place in the base, prior to final installation. High frequency on the left, low frequency on the right. Input is on the right (the rear base panel). I used compression connectors to connect speaker wires to the crossovers.

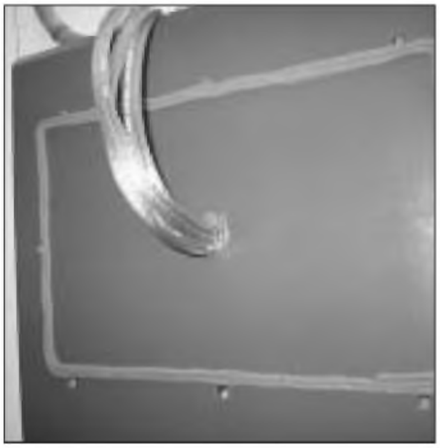


PHOTO 13: Top of the base. After taking the photo I removed one row of Mortite to avoid more overkill. Speaker wires are sealed with hot-melt glue.

PHOTO 14: Quilting material spread out and being rolled into a width for V1 cavity in the T-line.

and you want a tight fit between the tower and the base. I drilled ⁵⁄₁₆″ holes in the base top to give me a slight leeway for bolts in installing the base later.

Assembling the bases with biscuits was quite easy and went quickly (*Photo 10*), although I dry-fitted both before gluing and clamping. I installed the tops of the boxes next. I checked the internal baffle's top edge and added generous amounts of plastic wood before installing the top. You'll need four pipe clamps for this gluing task, and a couple of shorter clamps to make the front edge snug with the back of the front panel.

As soon as these were dry I upended the cabinets and matched the bases to the tower bottoms, carefully marking the box's bottom edges through the drilled base holes. It is very important to mark these accurately and to drill the holes at exactly 90° to the sides. I took the precaution of installing two of the brass inserts and temporarily bolting the base to the TL, and then drilling shallow pilot holes in the cabinet sides for the remainder of the holes to make sure they were exactly where they should be.

Installing the large brass inserts must be done by hand with a very large straight-bladed screwdriver (*Photo 11*). I discovered that adding a little Teflon® spray to the inserts assisted wonderfully in seating them. The McFeely's catalog shows you how to make a handy driver for the inserts using a short bolt and a pair of washers. The insert tops must sit just below the surface of the edge, with any bulges in the MDF sanded off carefully.

MOUNTING METHODS

If you prefer a way to easily remove and replace drivers, you could use #8 threaded brass inserts for the six mounting holes on the mid woofers and the four for the tweeter. If you do not round the inner edges of the driver holes, you can use the standard T-nuts to mount the drivers. Installing brass inserts is best done before any finishing of the cabinet.

If you do use this method, install two locating inserts for each unit and, using #8 machine screws, temporarily mount each driver, and then drill pilot holes for locating the other inserts. Do this very carefully, of course, to avoid damage to the drive units. I believe this is the only method which will locate the inserts accurately. If you use threaded brass inserts, cover them with masking tape if you plan to paint the enclosure.

FINISHING

Rarely is a cabinet assembled which has perfectly mating edges. For this situation, the belt sander was invented. Keep clearly in mind, however, that MDF is very soft and a belt sander can remove a lot of material very quickly. Use the sander judiciously to even up joints that do not exactly match.

If you are so unfortunate as to have some voids in the joints, Plastic Wood will repair the problem. Sanding will level such patches. A smaller palm sander with #80 paper is next needed to produce a uniform surface.

At this point, after the remedial sanding, do some tests on scrap stock with your router using its ⁵⁄₈″ roundover bit, then round the edges of the front panels and the two side edges of the tops. The trick is to set a plunge router depth so it accurately rounds the corners without a ridge. Next round the top and corner edges of the bases. Do all these cuts at the same time to ensure uniformity.

I considered veneering, which I've done before with pleasure. But on these cabinets, the rounded edges were beyond my skill and experience. I took the easy (?) way out and decided on paint, a nearly black hue, which is a huge success with my women friends. I vacuumed all surfaces thoroughly and removed any residue with tack cloths. I was advised to do at least five coats—two with an oil-based primer just a shade or two lighter than the finish color.

I sanded after each coat, of course, using #220 grit on the palm sander for the first two coats, using tack cloths after each sanding. I discovered that a leather hole punch works fine to punch the appropriately located dust escape holes in each sandpaper quarter sheet. I made an exception on tool quality in buying the inexpensive Chinese punch, an exception proving my rule.

After each of the finish coats of acrylic water-based paint, I used #400 paper to develop a velvety finish on both the towers and the bases.

CROSSOVERS AND WIRE

Before building the cabinets I had assembled the crossovers. My assembly layouts in the original article worked out well. I used cable ties and appropriate ⅛″ holes to fasten components to the ¼″ × 5 × 8″ composition boards (*Photo 12*). I press-fit mounted silver-plated studs on these boards for input and output terminals. These were old and it took some effort to get rid of the accumulated oxide. I mounted these in the bases and added screw compression connectors for the outputs from each board.

I cut three appropriate lengths of wire for each cabinet, long enough to reach the three drivers with enough slack to change them if necessary. After attaching these to the crossovers, I threaded these through the hole in the center of the base (*Photo 13*), sealing both sides of the hole with hot-melt glue. Mark the tweeter wire with tape.

Stuffing comes next. Do not trust whatever weight the vendors claim if you buy quilting acrylic or pillow acrylic from the big chains. I like the quilting acrylic since it's easier to weigh and to install. I used a cooking scale to measure the two quantities for each cabinet.

I laid out the quilting material (*Photo 14*) and refolded it to a width matching the 7½″ internal width of the line. I was surprised at how easily both the quilting and the pillow stuffing slid into place in the two cavities. Be sure to take great care in weighing each batch of the stuffing materials to achieve ideal performance.

BASE TO LINE

To begin final assembly I placed each tower on its face on carpeting, first putting a scrap of 1″ stock covered with a soft towel to protect the finish under each end of the front panel. This made it easy to mate the base to the tower, thread the wires to their appropriate holes, and to install the first two bolts. I put one strand of Mortite® just inside the holes in the top of the base, making sure I had an unbroken ring that would seal the bottom of the T-line completely. You could also use ⅛″-wide adhesive insulation foam for this task.

Once the first two bolts were loosely in place, I upended the assembly to complete installing the remaining six bolts. Tighten these uniformly to compress the Mortite, checking the join as you go. Do not over tighten since the threads embedded in the MDF could be pulled loose.

Turn the assembled cabinets over and trim the wires to a length which allows you to install the drivers comfortably. Unfortunately, the driver connectors are metric-based, male spade lugs, gold-plated. U.S. female lug connectors do not fit these, and in any case such connectors are not readily available with gold-plated surfaces.

So tin each wire end thoroughly, bend it into a tight "U" and wrap it around the terminals, being careful to observe polarity. A soldering gun is easiest to use here to heat the wire and lug quickly and thoroughly.

I used a #14 gauge wire for this internal wiring but was amused to note that the tweeter's wires from the terminal to the voice coil look like #36 gauge. The nice Litz wires in the mid-woofer units are larger, but far smaller than the #14 I soldered to their lugs.

I used #8 × ⅝″ Phillips round-head stainless steel metal screws to mount the drivers, hoping I don't have much occasion to remove and replace them.

POST PARTUM

As I write this the THORs have been in place in my listening room for only two weeks, too short a time for the drivers to have really been broken in (*Photo 15*). I am also still experimenting with which amplifier to use with them, and in what arrangement.

At present I have two, two-channel Pass Zen amps in place with each of their two channels simply attached to the treble and bass terminals. The treble is a bit too prominent, but I'm waiting to see how the break-in period may change them. Since the Pass amps may be bridged, I will parallel the crossover inputs to see how this topology behaves. I have a couple of stereo Adcom power amps which I plan to try, and my old Brute, stereo 70 amp, which is being refurbished. I may try combinations, as well.

How do they sound? Well, trying to avoid the hazard of the builder's prejudice, I'll say they sound as good as Joe D'Appolito's pair, which I auditioned with Dennis Colin and Joe in the latter's listening room. Their range is impressive, and especially their ability to image. Others who have listened to them are deeply impressed by the illusion of the presence of the performers. This is especially true with small instrumental groups and vocalists.

I really have a hard time thinking of a proper analogy for what it feels like to have built a pair of loudspeakers. I was tempted to mention child-bearing, but quickly abandoned that, just for the sake of prudence. Like many unique experiences, however, unless you have done this sort of thing yourself, you have no idea of the resulting emotional high. There are some things you cannot know about loudspeakers in any other way.

When you can finally put your handiwork in place in your listening room and settle into the sweet spot chair, you put on a recording which you know is exceptional and relax. In-

PHOTO 15: The finished THORs in my listening room.

evitably your left brain, which has been overworking for weeks on this project, slinks over into a corner and the right brain takes over. It takes quite a while for the goose bumps to subside, the wonder and surprise to moderate, and a fair judgment to balance your assessment. Fortunately, I had a benchmark in having heard the original prototypes. Since completing my pair, I have been listening carefully for changes as they break in and am going through my music collection doing comparisons.

The THORs are very analytical and the differences in disk quality are more evident than I would have believed. I am coming to believe that their ability at imaging is better than any of my previous units. The depth is good, and I believe it is improving.

FINALE

Are the THORs worth building? I firmly believe they are. The rewards are very great, including the satisfaction of having made the unit yourself. The drivers are some of the finest available today and are obviously technically sophisticated. (Refer to the original article for a technical description.)

If you build a pair I firmly believe you will be deeply impressed by the transmission line design and their truly outstanding performance. If you like building things, assembling a pair of THORs is a great adventure, and one I would not have missed.

You can, of course, save a lot of hard work by ordering Madisound's complete kit for $1550 plus shipping (608-831-3433, http://www.madisound.com). It includes spikes and, unlike the prototype, has no base. It also includes a grille. SEAS drivers and suitable crossover components are available from a number of vendors. ❖

A 6L6G Parallel SE Amp

Put your collection of old audio tubes—6L6Gs, GCs, or GAs—to use by building this 6L6G PSE (parallel single-ended) stereo amp running in triode mode. By Mark Weaver

This amp (*Photo 1*) might not sound quite as good as the 300B amps I build, but it's close, and if you don't have 6L6GA tubes in your collection, you can purchase them for only about $14 each as NOS (new old stock). Try finding NOS 300Bs for 14 bucks! I also like the old coke-bottle look of these tubes. A couple of 6L6GAs in parallel sound surprisingly loud through my speakers. I'm still having a difficult time believing it's only about 3W per channel. Svetlana is currently manufacturing 6L6GCs, a smaller version of the tube, which appear to be well-made and are not expensive.

PHOTO 1: The completed 6L6 amp.

PARTS SELECTION

All of the parts are off-the-shelf and most are available from Angela Instruments (www.angela.com) and Antique Electronic Supply (www.tubesandmore.com). Everything fits on a Hammond 17 × 10 × 3 steel chassis. The output transformers are One Electron UBT-1s, which are of good quality, inexpensive, and designed for parallel output tubes. All of the other transformers are by Hammond.

If you prefer to use all Hammond units, the 1627SE output transformer is also a good inexpensive choice. I am certainly impressed by the size and weight of this SE output transformer—11 lb!

A rather hefty power transformer is required for this project because the amplifier draws 225mA. The Hammond 278CX rated at 465mA fits the bill—plenty of current but too much voltage for 6L6Gs running in triode mode. This power transformer doesn't have a 5V winding, so you need a separate transformer for the filaments of the rectifier tube.

A simple solution to the high voltage problem is to use a choke-input filter instead of the capacitor-input filter you typically see in tube amps, since a choke-input filter puts out less voltage. This was a good turn of events because I really like the sound of a choke-input filter and plan to stick with this design whenever possible from now on.

The British-based audio brand Border Patrol has produced a white paper on the use of choke-input filters in tube amps. Check it out at www.users.globalnet.co.uk/~valveamp/BorderPatrolWP.htm. The paper basically says that the choke-input filter provides the best possible regulation and "chokes" high frequency noise out. Other benefits include reduced core distortion and saturation in the power transformer. The resulting sound is more fluid and clean with a more natural timbre.

The power-supply choke I selected was another big hunk of Hammond iron—the model 193M rated at 10H and 300mA. But after finishing the power supply and slapping on a resistive load, my glee at all this iron turned to gloom—the choke buzzed like the dickens. A choke-input filter puts a lot of stress on the choke and it was vibrating like a woofer plugged into an AC outlet. If I were building a ham radio station the noise wouldn't bother me. In fact, I'd probably like it. But this was too much noise for a hi-fi amp.

CHOKE FILTER

A simple solution to the buzzing prob-

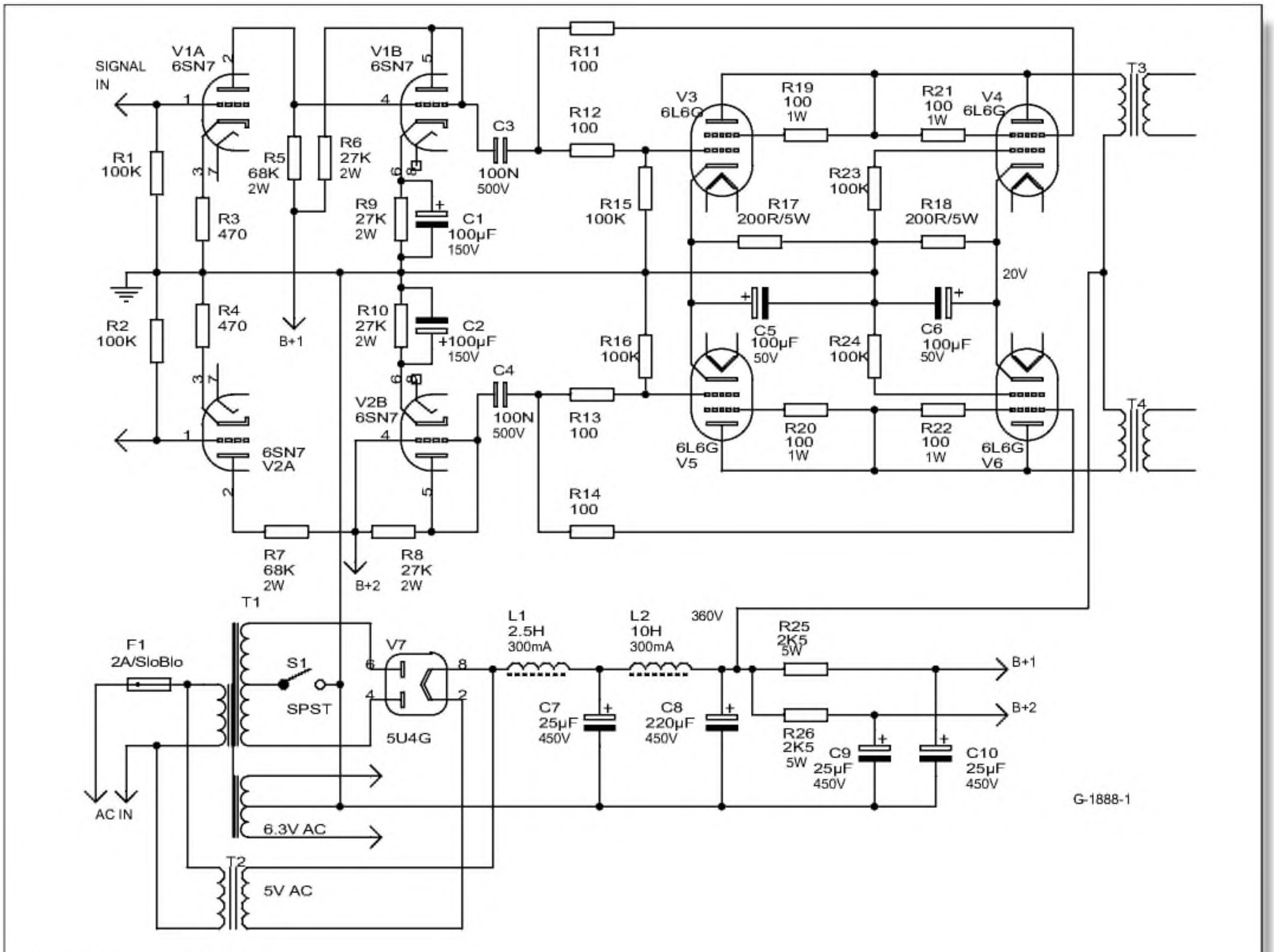


FIGURE 1: Amp schematic.

lem was to use two chokes instead of one (*Fig. 1*). A smaller-value choke as the first choke in the filter, followed by a 25µF bypass capacitor, took the edge off the DC ripple and eliminated about 80% of the buzzing from the big choke.

The smallest value that will work in this circuit must be at least what's termed the "critical value." This is easily determined by dividing the output voltage of the power supply by the total current drawn in milliamps. For this amplifier, it is something like 310V divided by 225mA, or about 1.5H. If the choke is not at least that value, the filter will act as a capacitor-input filter and the voltage will be much higher.

I ended up using a 2.5H choke as the first filter choke followed by the big buzzer—the 10H choke. The number-two choke doesn't need to be that big—around 5H should be fine. Why not be the big-boss-hog and design your own? Just make sure the first choke is at least the critical value. Also keep in mind that choke-input filters need a large final filter capacitor—at least 200µF.

For all I know, this might be the only tube amp with a double-choke filter. I don't think it is used much in audio equipment, but you can find it in the *Radio Amateur's Handbook*, for example. Or, if you prefer, use a single choke. I found adequate hum reduction in this amplifier by using a single choke as small as 5H.

OPERATION

The 5U4G tube rectifier warms up rather quickly, so this power supply slams the filter capacitors with over 500V for a few seconds before the 6L6Gs tubes warm up. So I used a commercial time-delay relay switch to delay the high voltage. A company called National Controls Corporation (www.natcon.com) makes these neat little epoxy-encapsulated time-delay relays that are 2″ square and are held to the chassis with a single bolt. I selected model Q1T-00060-341, which has an adjustable delay from 3 to 60 seconds and operates from the main AC line. You can order it from MSC Industrial Supply Company (www.mscdirect.com, 1-800-645-7270) for about $25.

The cheap, easy way to do this would be to use a toggle switch instead of a time-delay relay. But my wife would never understand why she needs to flip two switches, about 30 seconds apart, to turn on the stereo.

The input stage is a single direct-coupled 6SN7 in each channel. This gives plenty of gain and places only one capacitor in the audio path. This stage is followed by the 6L6Gs in parallel.

Don't leave out the 100Ω resistors connected to the control grids of the parallel output tubes or they may oscillate. I used 100Ω, 1W resistors to connect the

(to page 69)

Build Your Own Axisymmetric Tractrix Horn

This veteran builder shows you how to construct a full-size axisymmetric horn geometry with tractrix curve sound-wave propagation.

By Robert Roggeveen

Articles in *Speaker Builder*[1] have given historical perspective of the development of the tractrix horn (*Photo 1*), starting with Christiaan Huygens, who laid the mathematical groundwork for the curve. Most valuable are the interviews and reflections of contacts Dr. Bruce Edgar had with Mr. P.G.A.H. Voigt[2]. Dr. Edgar has also offered several tractrix horn building projects in these pages, most of which I have built.

Since these rectangular cross-sectioned wave paths are approximations of a true fluid line curve, I wanted to investigate whether a full radial cross section of a wave path could be made using easily obtained materials at low cost. I was driven in this search by the belief that such a horn would have different tone spectrum dispersion and would still be appealing.

I had read Voigt's explanation of the layout of a tractrix curve on graph paper. I copied the graphic layout as presented in *SB* 3/81[2] and then made a mirror image of it. I placed the copied image opposite the mirrored image to form what looks like a cross section of a horn as viewed from the side. The radial lines remain the same lengths as they move into the horn towards the throat.

Now, if each line pair is a cross section of a cone, then by constructing a series of cones with ever-increasing pitch angles and stacking them on top of one another I would produce a curve as shown on the graph paper. By using casting techniques, I can then transpose this curve onto a solid matrix, inverting space in the process—that is, inverting a solid into a void and conversely a void into a solid. Marie Shrewsbury[3] gave an example of casting a concrete horn and provides valuable information in her article.

The photographs that accompany this article show various stages of building four different horns. All four use the same process and procedure, the only variant is the diameter of the disks used to build them. So read the text as a reference and use the photographs as a general visual aid.

PHOTO 1: Author with axisymmetric tractrix horn.

THE CONES

The horn in this experiment mates to the Siare 16 VR, and is compared to the Edgar midrange horn[4]. The mouth should be about 14″ in diameter. I oversized the radius by 2″.

Cut 28 disks with a diameter of 18″ out of construction paper (*Photo 2*). Use the first disk you cut as a template for all the others. I used two kinds of paper, one thicker than the other. The tensile strength is higher in the 200 lb paper, which will resist buckling under the weight of the concrete. I used this heavy paper to make the first 12 cones.

Higher up the neck tensile strength is less critical, so you can use a lighter weight, say, 80 to 100 lb bogus paper. You can find these in your local arts and craft store or through Nasco, 800-558-9595.

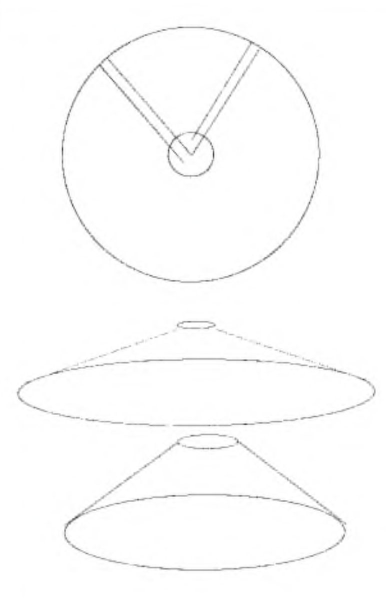


FIGURE 1: Measuring and cutting the cones.

In order to glue the paper you must conceptualize the glue strip and make allowances for it as you cut the segments (*Fig. 1*). I lessened the first disk by 1/24 as measured along the circumference. I drew that piece of pie. Now draw a line ¼″ parallel to both long sides of the pie shape, both on the right side.

The space between these two narrow strips will be superimposed on one another and glued with rubber cement. So cut out the segment. Gluing is made easier by cutting a small circle out of the center of the disk; say 2″ in diameter (*Photo 3*). Apply rubber cement to the glue strips, let it become tacky and stick them together.

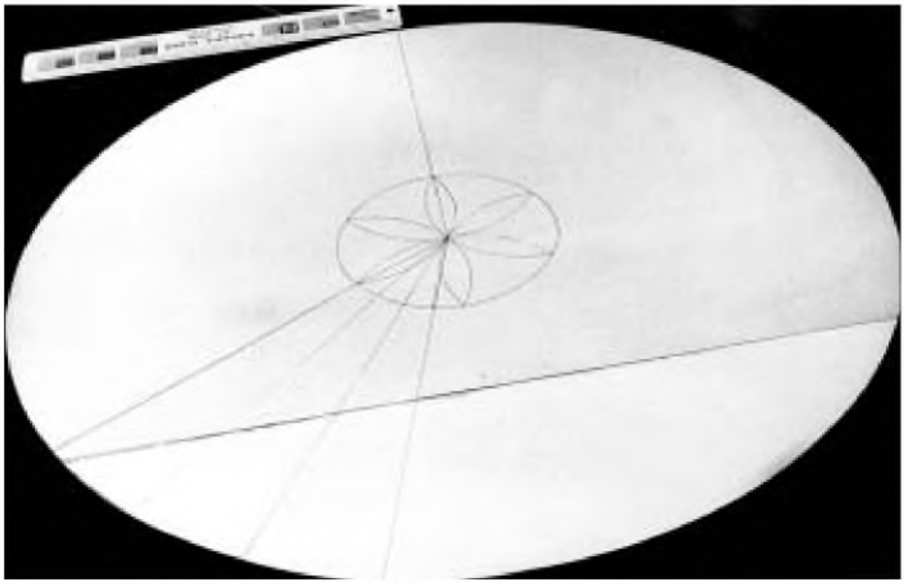


PHOTO 2: Fashioning the 18″ disks from construction paper.

This is the first cone and it has a shallow pitch (*Photo 4*). You make the second cone from a disk with a circumference lessened by 2/24. Cut and glue it the same way as the first. Make the third cone from a disk with a circumference reduced by 3/24, the fourth by 4/24, and so on until 20/24.

You will use the other eight disks to fine-tune the shape of the horn by making smaller increments and placing them in the sequence. I made 3/48, 5/48, 7/48, 9/48, 11/48, and 13/48 reduced cones. In fact, the more cones you use—or, conversely, the smaller increments you use—the more accurate the tractrix curve will be represented by the mold.

I made my first horn using 1/24 increments (*Photo 5*). It has a mouth cut-off of 400Hz, which is 12″ in diameter and a throat of 2¼″ diameter. This compares well with the long horn of Edgar's midrange. However, my concrete result is shorter by a couple of inches (*Photo 6*).

By using more cones the horn length will increase, more closely approximating the tractrix curve. To be practical I used 30 cones, but you can use more, obviously.

THE MOLD

I placed the stacked cones on a paper base disk with a radius 2″ longer than the disks used to make the cones. This base disk will form the flange of the horn. Start with the first and biggest cone, 1/24, and place it with the large circumference touching the base disk. Center the second cone, 3/48, over it.

As you will note, I incorporated the "fine-tune" cones (*Photo 7*). Place the third, 2/24, on the second (*Photo 8*) and so on, each time centering the cone (*Fig. 2* and *Photo 9*).

Next glue them all together. I used epoxy because it also seals the paper from moisture. Extend the glue onto the flat base disk.

At a right angle make a paper dam along the edge from a strip of paper (*Photo 10*) cut about 1¾″ wide and 16″ long. Next draw a lengthwise line 1″ from a side. Then cut a sawtooth edge on the other side making teeth of ¾″ deep. Fold these teeth under the base disk and glue with rubber cement. You will need more than one strip to go all the way around the perimeter of the base disk. Then paint the dam on the inside with epoxy.

THE DRIVER PLATE

The diameter of the throat is equal to the diameter of the driver's cone, about 4¼″. The Siare 16 VR is mated in a 1:1 ratio to the horn. Draw a circle with a 2 [illegible]/8″ radius on a piece of ¾″ plywood. Next draw a larger circle, using the same center point, and producing a ring about 2½″ wide. Now cut out the ring, keeping in mind that the inner circle has a slight pitch relative to the pitch the horn has at the throat. I used a fine-tooth saber saw for this.

Place the ring over the top of the mold. It can slide only so far until it rests on the mold (*Photo 10*). Make sure the ring's plane is parallel with the base disk. Draw a line on the mold at the underside of the plywood. Take the ring off.

This is a good time to decide on your method of mounting the driver, the Siare in this case. I suggest T-nuts, because regular wood screws wear out the plywood ring. You could use other methods available, such as brass inserts. The point is, don't use screws.

There is one more reason to use durable bolting mechanisms. If at a later date you choose to change drivers, you'll want to be able to install an interface plate. T-nuts or other such methods will allow for that.

Now back to securing the driver plate to the concrete. Screw eight short screws into the underside of the plywood ring, roughly along the outer third of the ring's surface. The screws anchor the wood to the concrete. I predrilled the holes, being careful not to penetrate the plywood all the way through. I used 1½″ screws drawn ½″ into the wood.

It is a good idea to know where these screws are. I placed the screws along the lines of the T-nuts and halfway in between (*Photo 11*). When the horn is finished you have the mounting bolts to refer to where the screws are. Applying this thinking makes your horn ready to experiment with at a later date.

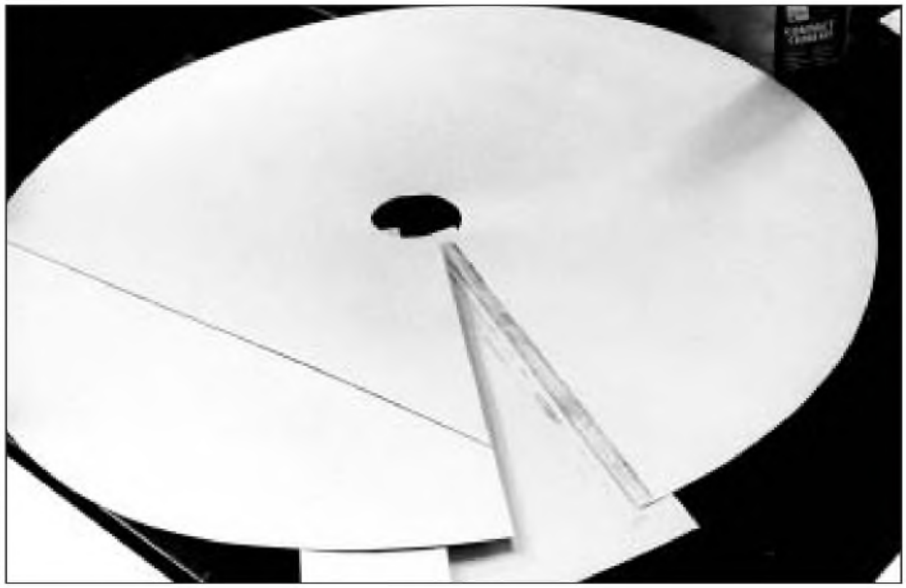


PHOTO 3: Cutting the first cone.

PHOTO 4: Showing the shallow pitch of the first cone.

PHOTO 5: The author's first horn (12″ diameter, with 2¼″ diameter throat).

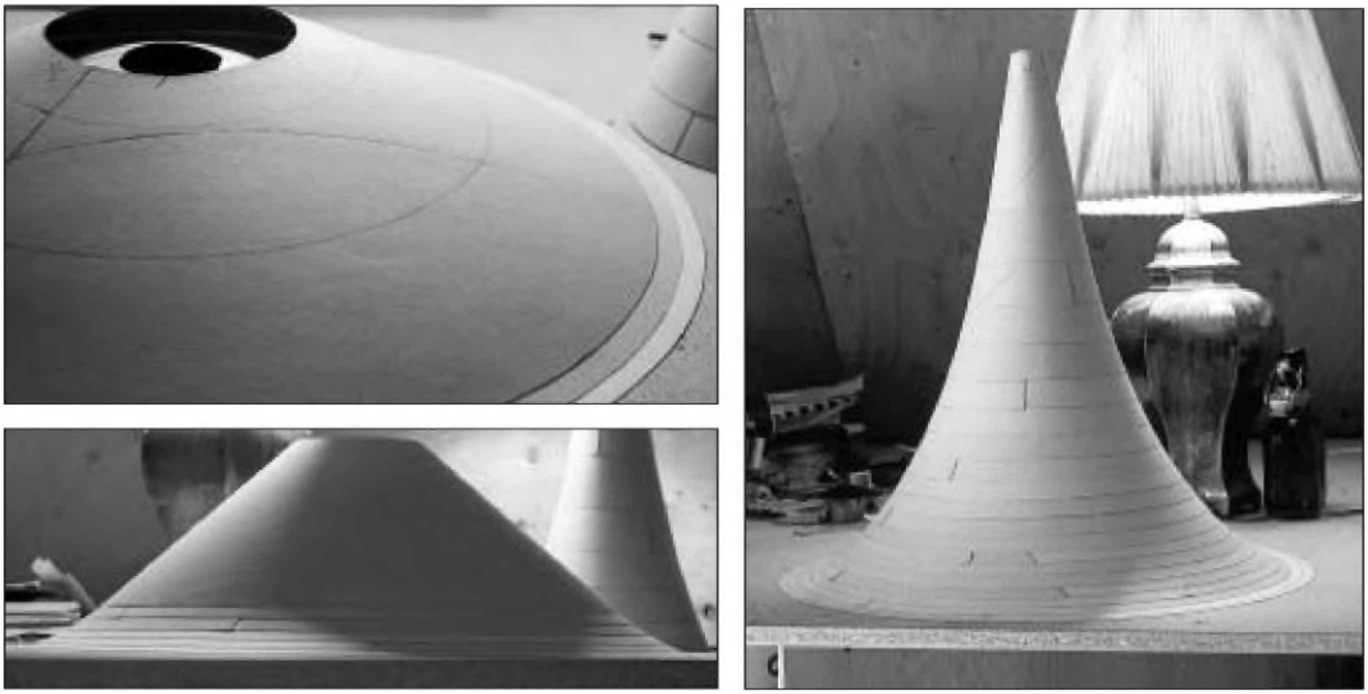
PHOTO 6: Comparing horn length with Bruce Edgar's midrange.

THE CONCRETE

Mix mortar, not too wet, and lay it on the mold. The concrete I used is a ready mix mortar made by Oldcastle Type S. It is used to lay brick and has no small rocks. It is also of high compression strength.

Use frequent chopping motions into the mortar so that it bonds well with the previous scoop you laid, and work the air bubbles out. Working methodically will produce fewer voids on the inner horn surface. Work at a steady pace and thoroughly from the base edge going around and slowly up the neck of the horn, applying mortar in even thickness, about 1″.

If you have not worked with concrete before, you may prefer to experiment



PHOTOS 7–9: Horn takes shape from paper cones.

and get your bearing. Possibly there are alternatives to concrete, such as "sculp-ta-mold," a papier-mâché material that I have not tried.

At the top of the neck just past the circle you drew, stop applying mortar and even it out by gently tapping the mix from the top with the ring you made. Go as far as the circle you drew, keeping the ring parallel with the base plane. It helps a lot if you can walk all around your mold, hunkering down, eyeing the ring as to its level.

When you have finished, slightly rotate the ring and take it off, laying it aside. Place about six nails halfway into the mortar near the throat. I used 1″ tack nails. Place them laterally; they act like rebar between the first and second mortar applications. Let the mortar dry.

THE HORN

Take out the paper mold. There is no elegant way to do it, and it pains me to destroy such a beautiful mold. The notion to make a mold that could be re-

PHOTO 11: Readying plywood ring for mounting.

PHOTO 10: Gluing the cones. Note paper dam at base.

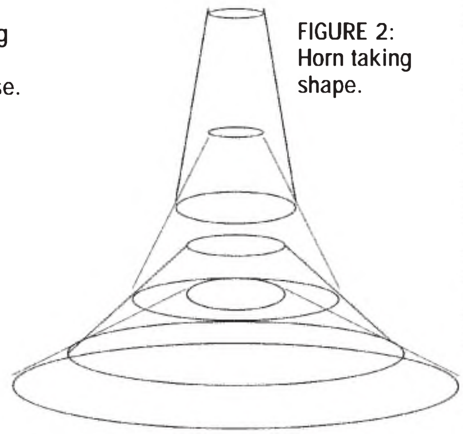


FIGURE 2: Horn taking shape.

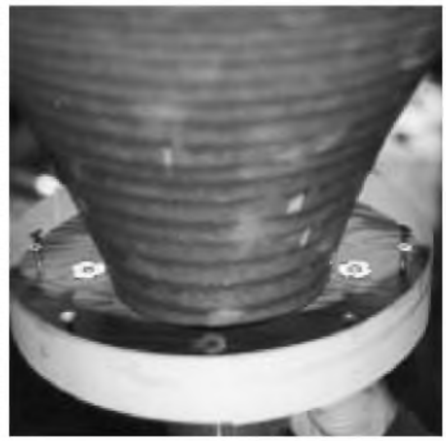


PHOTO 12: Paper edge around driver plate.

used is heavy on my mind. In some ways, you'd need to start with a different building process, for example, a wooden mold, sanded smooth. Or a clay mold, which is more difficult to make truly symmetrical.

When all stacked on one another, the paper cones produce a curved surface that has small steps. Looking closely, you see each additional cone make a tiny ridge. This is why it is impossible to just pull out the prepared paper mold. It acts like a fishhook. Still, this project is very cheap, under $20 and highly rewarding.

PHOTO 13: The horn finds a home hanging from the ceiling.

Once you have torn out the paper mold, you are left with a concrete horn shape. Form a paper edge around the driver plate using a 2″ wide strip of paper (*Photo 12*). The driver plate is ¾″ thick, so the 1¼″ left will stick out on the side of the anchoring screws. Staple the paper to itself at the overlap.

Fixing the driver mounting plate onto the concrete throat terminus takes a separate mixing of mortar. First, make a "dry run" of how things come together. The wooden ring lies on a flat surface, screws facing up. Place the concrete horn throat on to it with the expansion reaching straight up. It is quite precarious, but it works.

You may need to rotate the ring relative to the concrete a little to find the best match with the screws and nails and centering the hole. The screws are meant to anchor the ring onto the second mix, which you apply outside of the throat and onto the wooden ring. The thickness of the concrete horn is about 1″. The ring is about 2½″ wide, so that leaves 1½″ outside of the throat terminus. That's the space that has the screws in the wood, and as you apply mortar they will become embedded.

Once you have the right fit, make a mental note or a pencil mark to orient you. Now you are ready for the real thing. Lift the horn off the driver plate. Plug the bolthole of the T-nut with some grease—Vaseline® works well. This will keep the concrete from fouling the threads. Using a bonding agent, paint the wood surface as well as the throat terminus and the outside of the horn's neck to a point where you know the mortar will be applied. I used Quikrete® concrete bonding adhesive.

Place the horn back on the plate as noted before. Mix mortar and slop it into the moat you produced finishing the top surface in a slanting fashion upward, towards the neck of the horn. You will have embedded the nails and screws. It pays to use a thin spatula to work the mix around the screws, into the corners of the area. Let it dry for six hours. You can now carefully take the

PHOTO 14: 36″ diameter tractrix horn.

2″ paper strip off. Let the rest dry overnight.

I mounted the Saire 16 VR the same way Edgar did, with a ½″ gap, using nylon spacers found at the hardware store.

HOW IT SOUNDS

The sound is distributed more evenly into a broader space. It also is louder than the Edgar horn. I am not noticing a ringing of the bell shape of the horn, although it clearly has one when you rap your knuckles against it. It makes for a nice complement to the monolith Edgar bass horn[5] and EV tweeter. Rather than making a stand, I hang the horn from the ceiling (*Photo 13*). I am very happy with this sound.

I have since made a 36″ diameter tractrix horn (*Photo 14*) mated 1:1 to a 12″ EVM-12L that I had on hand from building the Edgar Show Horn[6]. I play this without a crossover. It has a significantly broadened midrange spectrum. Since these are full-size horns you could place them in a wall, thereby lowering the lower cutoff further. I have not tried this, but you may want to.

The electronics I enjoy are a pair of 60W Borbelys[7] driven by Boak's regulated power supplies[8] built in 1985, and a Carver preamplifier/tuner CT-26V. I listen to LPs mostly on a Technics SL-D2. I have another pair of 60W Borbelys, which I use for the rear channels of the surround-sound experience with video. I think this horn experience is coming close to that.

POSTSCRIPT

Since submitting this article for publication, the following marvelous experience happened. On a recent grand car tour, my wife Susie and our dog Finn traveled with me to (among other places) Hope, Ark., home for more than 50 years of the Paul Klipsch loudspeaker plant. I had called there in advance, and Cheryl Williams expected me as planned to tour the Paul Klipsch Museum of Audio History at the end of April 2002. There you can touch and inspect horn models that go back in time: Victrola, Voigt, RCA, and naturally numer-

PHOTO 15: Testing the author's prototype at the Klipsch factory.

FIGURE 3: Frequency response.

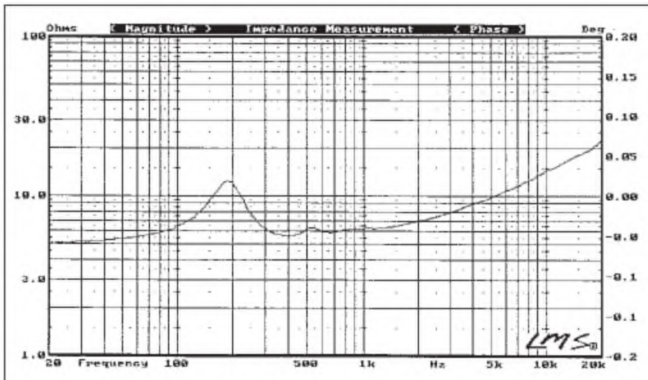


FIGURE 4: Impedance plot.

ous varieties of Klipsch horns—some that were in production and several that didn't make it.

I was in horn heaven, and Cheryl and Christopher Williams were graciously spending all afternoon with us. We toured the laboratory and saw the Jubilee, Paul's last design, which will be in production next year, I think. I had brought my prototype, 2¼″ throat, 12″ mouth, 9″ length (*Photo 5*), with a 19mm JBL 104H-2 driver flush-mounted to it. It was tested in the anechoic chamber, which was exciting and quite an honor.

Driver voltage was 2.71V, microphone distance was 3m (*Photo 15*). The LMS sweep gave the following profile of the frequency response (*Fig. 3*) and impedance plot (*Fig. 4*). Chris tells me that at one meter the reading will be 6dB higher. "A nice little horn," he said. A more pleasant affirmation could not have been hoped for: to have your own design tested and appreciated by the staff of the Klipsch factory!

So this experience has reaped many rewards and memories. The ultimate would be to be hired on! I haven't given up. Happy hobbying with and great listening to horns. ❖

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8. Boak, James E., "A Family of Power Amplifier Power Supplies," *TAA* 1/80, p. 6; "Power Modifications for ST-150-BJ-1," *TAA* 2/81, p. 40.

Followers: Cathode, Plate, and Others

Some of electronics' most useful circuits—tube or transistor—are ones where output "follows" input, in phase. They change impedances, isolate circuits . . . **By Norman H. Crowhurst**

This article appeared originally in Radio-Electronics, *May 1966. Courtesy of Jack Boyle, Southampton, N.J.*

The cathode follower is an old favorite, and you occasionally hear about plate followers and transistorized equivalents of these circuits. The word "follower" originated to explain the action, but sometimes its implication gets taken too far. So let's look at the whole group.

FEEDBACK REALITY

In the cathode follower, cathode voltage is supposed to follow grid voltage. Where the source of grid voltage applied may have a high impedance, the duplicate produced at the cathode provides an abundant supply of current at low impedance (*Fig. 1*). That's the notion. Another description of the cathode follower says it has 100% voltage feedback. And because it has 100% feedback, it is assumed that the waveform must be perfect.

. . . [But], that "tain't necessarily so." That 100% feedback statement means that the feedback fraction, beta (β), is unity, or 100%. The erroneous notion follows from confusing that beta with the feedback factor ($1 + A\beta$). If the latter were infinite, gain would be 0dB, and the voltage-gain-with-feedback expression,

$$\frac{A}{1+A\beta}$$

would have to equal one.

Making this substitution and doing a little algebra, to solve for beta, you find the feedback fraction must be

$$\beta = 1 - \frac{1}{A}$$

which is quite realizable in theory. To achieve it in practice, if the gain of a tube, with a particular plate load and operating point, is 50 (ratio, not dB), beta needs to be 49/50, or 0.98. Put 0.98 of the load in the cathode and the other .02 of it in the plate, and take the output from plate to cathode (*Fig. 2*), and you have no gain (unity gain, that is).

In practical use, the cathode follower's output is always taken from cathode to ground, which upsets the gain calculation of the last paragraph. Actually, gain of a cathode follower is always a fractional ratio, *nearly* one, or slightly negative in dB. In the previous example, the gain proves to be 50/51, approximately 0.98 (as a ratio), or –0.2dB.

PRACTICAL CIRCUIT

This can be applied to a practical circuit (*Fig. 3*). A bias resistor of 1kΩ sets the correct operating point for the 12AX7 with a 75kΩ plate load. This load is connected from the lower end of the bias resistor to ground. The circuit will have a gain of 50/51 (–0.2dB) and will reduce distortion by the feedback factor of 1 + 50 = 51. The source impedance presented at the output is the reciprocal of the tube's transconductance. The grid-return resistor, shown as 1MΩ, is effectively multiplied by the same factor of 51 in determining the circuit's effective input impedance (loading), and its grid-to-cathode capacitance is divided by 51.

If the tube's distortion at full swing,

PHOTO 1: These scope traces, made with emitter follower of *Fig. 10*, show the drastic effect of hanging a "matching" load onto the output. This applies equally well to tube cathode followers. Trace 1 was made with output open-circuited (unloaded) and a supply of 12V. There is almost no distortion, and gain is very nearly 1. Connecting load of 15Ω to match internal source impedance results in trace 2. Transistor is almost completely cut off except at left end of trace, where small amount of signal gets through. Distortion would be unbearable. Trace 3 shows what happens when input is reduced (with load still connected) so entire signal is amplified. Signal voltage has to be cut to 0.3 RMS—one-tenth of signal used in trace 1. Output was only 0.15V—indicating gain has dropped to 0.5 instead of almost 1. Curvature of trace 3 indicates severe distortion; opening of loop shows phase shift between input and output signals.

with the plate load used, is 5%, working as a follower will reduce distortion to about 0.1%. Assuming the transconductance is 1,250μmhos, the source resistance at the output is about 800Ω.

Now comes the hitch. Using the classic power-matching formula, the cathode follower is expected to work into a load equal to its output source resistance, while retaining its other advantages. (*Please* don't try to "correct" me in what I am about to say! . . . [Others] have quoted their college professors and misquoted textbooks such as Terman and goodness knows what to prove me wrong! If you don't believe what I say, *try it.* Set up the circuit and measure it for yourself.)

The cathode follower of *Fig. 3* has a gain of –0.2dB, *only if no load is connected.* It has a distortion of 0.1%, *only if no load is connected.* An output load equal to the cathode-to-ground resistor will reduce gain to about –0.3dB and increase distortion to about 0.15%. But put on a *matching load* and it's a very different story.

Couple an 800Ω load to a tube with a transconductance of 1,250μmhos, and working gain drops from 50 to precisely one. Not only that, but the distortion without feedback at full swing jumps from 5% to about 30% as a result of loading. Acceptable input swing is also cut drastically. The feedback factor is now 2 instead of 51, so follower gain drops from 0.98 to 0.5 (or from –0.2 to –0.3dB), distortion jumps from 0.1% to 15%, and the available output voltage drops by about 50:1, because the tube delivers only about the same *current* swing. If the unloaded cathode follower delivered 50V RMS at 0.1% distortion, the "matched" follower would deliver only 1V at 15% distortion!

Apart from the fact that loading destroys the advantage of a cathode follower, the normal limit to its working

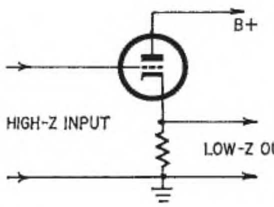

FIGURE 1: Basic cathode-follower idea. As the grid is driven negative, current through the tube decreases, as does current through the cathode resistor. That means less voltage drop across the resistor—so its top end gets closer to ground potential—more negative, like the grid.

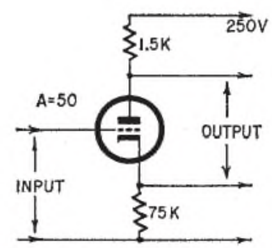

FIGURE 2: Circuit for theoretical unity-gain stage (see text).

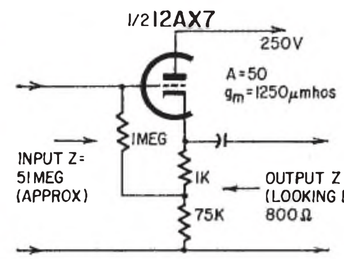

FIGURE 3: Practical cathode follower using 12AX7 with working gain of 50.

range is sometimes overlooked. You would think it's obvious that, with a 250V plate supply, a cathode follower could not handle a 300V signal, but the limits do get forgotten. Cathode voltage must always be a little positive of grid voltage, by the same amount the grid is momentarily negative from cathode (measuring the same voltage the opposite way).

The more positive the grid goes, the higher the tube current and the less the *momentary* cathode-to-grid voltage must be. When this difference reaches zero, the grid starts to conduct, as in any other tube circuit, abruptly clipping the waveform. At the other end, the grid can go negative of ground only by a voltage equal to grid cutoff for the operating load line chosen. *Figure 4* shows relative voltages at these instances.

From the fact that loading destroys the low-distortion advantage of a cathode follower, you can see just where the follower is and isn't practical. It is not, as has sometimes been claimed, an improved substitute for a line-matching output transformer—the line transformer does a better job, where *matching* is required. But for providing a *low-impedance source* to an output line—for example, from an FM tuner, preamplifier, or control center, either in a home high-fidelity or professional system—a cathode follower does serve better than a line transformer, *provided it connects only to a high-impedance input.*

[With certain transistor emitter follower circuits, it is possible to produce source impedances so low that they will work effectively into *load* impedances of 600Ω or less (see "Transistor Line Transformer," April 1966, *Radio-Electronics*). But this is still not *matching*, and the material in the preceding paragraph still holds. The load should not be less than 100 times the effective source resistance if the properties of the cathode or emitter follower are to be kept.—*Editor*]

PLATE FOLLOWER

Here plate current is supposed to follow cathode current, input being applied to the cathode, with the grid grounded. A larger voltage is available at the plate than is presented to the cathode (*Fig. 5*). This circuit is a little more difficult to visualize, because in practice you must provide somewhere for the cathode current to "go." But first let's treat it as an "ideal" follower.

Without any other connection to the cathode, plate current must equal cathode current, unless the grid is more positive than the cathode (cathode negative of grid, or ground). So, if a certain input current is fixed, plate voltage will be fixed by the drop in the plate resistor (*Fig. 6*). Cathode voltage is fixed as the equal and opposite of the grid voltage that would be required for this plate voltage and current combination. If cathode current is changed, the change in cathode voltage must be equal to the change in plate voltage, divided by the working voltage gain of the tube. This means the effective resistance in the cathode circuit, presented as a load impedance to the input source current, is the plate resistor divided by the working voltage gain of the tube.

This circuit is subject to the same limitations as the cathode-follower circuit at cutoff and at zero grid voltage. It has no negative feedback, so the voltage and resistance transfer ratio is as nonlinear as the tube's characteristics from the operating point chosen.

When you try to provide a constant-current input circuit to the cathode, while also achieving correct bias, a difficulty appears: the first requires a high input resistance and the second a much lower value of resistance between cathode and ground. The circuit can be used in its true form only in conjunction with a transformer, either tuned to a specific frequency or a wide-band audio type (*Fig. 7*). The impedance presented by the transformer to the cathode must be high enough to achieve essentially constant-current input, while the winding resistance (padded, if necessary) between cathode and ground sets up correct bias.

For this reason, the plate follower has found little use in audio circuits. It has proven useful in higher-frequency radio circuits (usually under the name "grounded grid"), where it helps keep unwanted circuit interaction at a minimum, by using lower working impedances throughout than do other tube circuits. The *cathode* follower provides a low output source impedance, but a

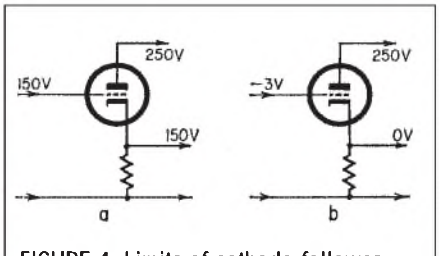

FIGURE 4: Limits of cathode-follower action, for circuit of *Fig. 3*. In (a), grid is driven positive until its voltage equals cathode's. Then the tube saturates and positive signal peaks are clipped off. In (b), grid is driven negative enough to cut off the tube, bringing cathode voltage to zero. Further negative excursions of the signal are chopped off. Note the tremendous signal "swing" available with cathode follower—0 to 150V at cathode. But grid must swing from –3 to 150V, which is a greater range than cathode signal.

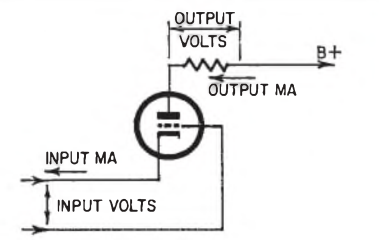

FIGURE 5: Basic concept of anode (plate) follower—also called "grounded-grid" stage.

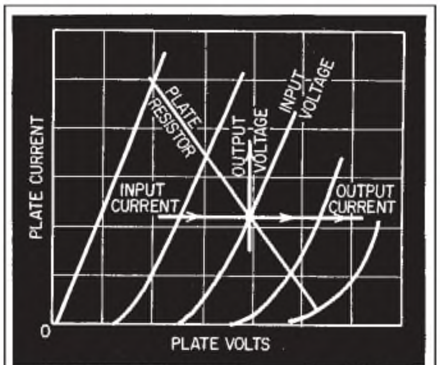

FIGURE 6: Plate-follower relationships shown on load line.

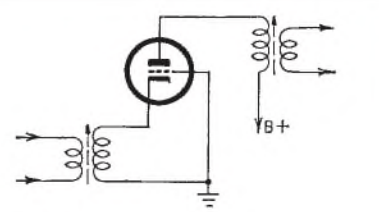

FIGURE 7: Practical plate follower is very simple, if DC resistance of input winding is large enough to produce sufficient grid bias.

very high input loading impedance. The plate follower loads down its input source and the output impedance is no higher than conventional plate circuits. Thus it serves in the nature of an impedance stepup, at constant current, so voltage gain is the same as impedance stepup effect.

TRANSISTOR VERSIONS

The transistor version of the cathode follower is an emitter follower (*Fig. 8*). Bias is simpler for an emitter follower than for a cathode follower. But the result has certain differences.

In the cathode follower, the source resistance is governed by the tube's transconductance. In the emitter follower, the relevant parameters are working-current gain *ratio* and the source resistance presented to the transistor. With the tube the source resistance the grid sees is almost completely irrelevant. With the transistor, base source resistance almost entirely controls the reflected source resistance.

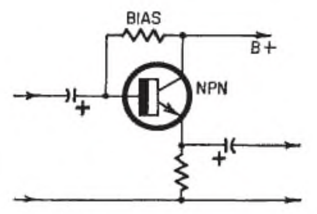

FIGURE 8: Basic emitter follower for n-p-n transistor. Looks and works much like cathode follower.

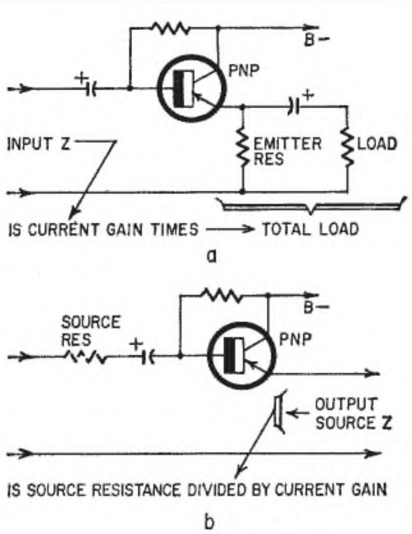

FIGURE 9: Impedance reflection in emitter follower.

Transistor gain varies along its characteristic, so the reflected source and load resistances (it works both ways—*Fig. 9*) will change at different points on the waveform, to some extent. However, the emitter follower is essentially like the cathode follower in being a voltage-follower device: the emitter voltage closely follows the base input voltage.

But in this respect its action is subject to limitations like its tube counterpart. *Figure 10* shows the circuit with which scope traces (*Photo 1*) were taken. With the output open-circuited, input and output voltages, using a 12V supply, measured 3V RMS (both almost identical, as shown by the straight-line trace 1).

Connecting the matching load (calculated at 15Ω here) limits the output, causing the transistor to be cut off most of the way, with only a little kink at the left end, where it still transmits signal (trace 2). Cutting down the input to where cutoff no longer occurs (trace 3), the input measures 0.3 and the output 0.15V. Note the curvature, which indicates substantial distortion.

The transistor version reflects impedances both ways, which the tube version does not. The input impedance presented by an emitter follower is its load impedance (including reactance, if any) multiplied by the current gain ratio (h_{fe}), plus the base-to-emitter AC resistance of the transistor (measured at constant collector voltage), which is usually relatively small. The output source impedance is the source impedance presented to the base divided by the current gain ratio, plus base-to-emitter resistance (measured with collector open), which gain is relatively small.

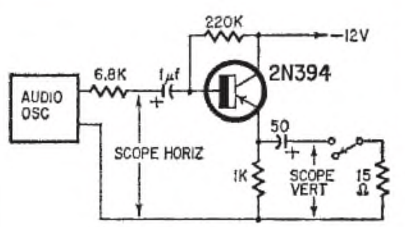

FIGURE 10: Circuit used in making scope traces shown in *Photo 1*.

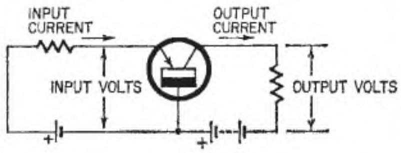

FIGURE 11: Transistor equivalent of plate follower is common-base stage.

The transistor equivalent of the plate follower is more basic to the transistor—it is the grounded-base configuration (*Fig. 11*). The difficulty of separating bias and input source impedance functions is not encountered, so the current-follower action can easily be realized.

Just as in the tube version, there is no inherent feedback, except for a small amount included in the transistor's internal parameters. Any nonlinearity of the transistor working in this mode is not reduced in any way by the so-called follower action. However, most recent transistor types have quite good linearity—better than tubes. The main difficulty with this configuration is that it requires supplies of opposite polarity for the emitter and collector circuits. ❖

Subwoofer Placement in Nonrectangular Rooms

This article examines corner locations of subwoofers and some pros and cons of single-woofer versus dual-woofer systems for a given placement within your listening room. **By Bohdan Raczynski**

Subwoofer placement is already a well-debated topic. Chances are, you have read (and experienced) quite a lot about golden rules and ratios, corner placement, exciting room modes, and SPL modeling in rectangular rooms. In doing so, you may have noticed that the great majority of publications stop short of tackling nonrectangular rooms[1]. Indeed, even most computer software modeling SPL in rectangular rooms does not go beyond this simple geometry. Image method is employed most of the time for this purpose, being recognized for simplicity of implementation[2].

So, what if your listening room (or space) does not conform to this simplification? Well, if this is your problem or your area of interest, you will find something here for you.

A SIMPLE SETUP

The simplest configuration of a subwoofer system consists of a single subwoofer and two wide-range loudspeakers. The subwoofer is typically positioned in the "widely recommended" corner location, and the two wide-range loudspeakers are placed to produce an acoustic stage image for the listener. "Which corner?" you may ask. Well, not all corners are equal, as you will see.

In my previous articles on loudspeaker placement[3,4,5,6], I emphasized the importance of understanding the location of the room's nodal lines and pressure maximum. Now I extend the analysis and actually generate the "room contribution" curves for a nonrectangular room. Here, I am talking about SPL of an "ideal" speaker with a flat frequency response from 0–200Hz. Two such subwoofers are placed in the room, and the resulting SPL is plotted up to 200Hz. Showing only the room contributions offers better clarity for visualizing the room modal response, because it is not obscured by the subwoofers' own irregular SPL.

THE BASICS YOU KNOW

As you know, all rooms (a cavity volume enclosed by walls) have resonant frequencies at which the SPL generated by a source can be quite large. The frequencies at which resonances occur (called modes) depend on the geometry of the room. At a resonance frequency the pressure pattern inside the room will consist of antinodes, where the pressure is maximum, and nodes, where the pressure is zero. In a room with hard (e.g., reflective) walls, the pressure will always be maximum at the wall or in a corner when the room is excited at any of its modal frequencies.

If the source is located at an antinode for a given mode, the room response will be greatest. If the source is located at a pressure node for a given mode, the room will not respond regardless of how powerful the source is. Some of us (myself included) take advantage of our listening room's acoustic behavior to improve the performance of our systems. Placing a loudspeaker or subwoofer against a wall or in the corner of a room allows the low-frequency modes of the room to enhance the low-frequency performance of the loudspeakers.

When you place the source at a node for that frequency standing-wave pattern, the maximum room response drops to zero at that frequency. Moving the speaker some distance away from the node restores (at least partially) the room's response. Modal patterns occur only when the room is being driven at a modal frequency.

At any other frequency, the pressure waves radiating outwards from the source reflect from the walls, but do not combine to produce a modal pressure pattern. As a result, there are no nodes and antinodes and the pressure can actually fall to zero at a wall.

A LEAP BEYOND THE RECTANGULAR ROOM

If I were to model a low-frequency loudspeaker generating pressure patterns in a simple, six-wall room, I could simply employ a closed form equation based on the summation of images. However, in a more general case, the room will not be a simple six-wall cavity[6]. In that case, I would resort to the Finite Element Method (FEM) to take advantage of its accuracy at low frequencies and its excellent handling of complex geometrical shapes[7]. *Figure 1* shows a floor plan of an "L-shaped" room with subwoofer 1 (Sw1), subwoofer 2 (Sw2), and listener (L1) located at nodes "186," "188," and "0," respectively.

I have also used "brick" elements to approximate my listening room's internal geometry for the purpose of producing the FEM mesh. The "brick" element has the following dimensions: X = 0.50m, Y = 0.611cm, and Z = 0.25cm. With these dimensions in mind, I will get 20 elements per 20Hz acoustic wave and only two elements to approximate 200Hz wave.

It is obvious that the accuracy of my model deteriorates as the frequency increases. I could have chosen smaller elements and readily improved the accuracy at 200Hz. This would be great, but the penalty would be increased calculation time and RAM usage.

It is probably worth mentioning that

although my analysis covers the frequency range up to 200Hz, I am mostly interested in frequencies below 100Hz. These are easier to discriminate from the SPL modal plots and have more defined pressure patterns. Additionally, the low-end modes are more widely spaced, and therefore easier to deal with without affecting the immediately adjacent modes.

As a starting point, I have determined room modes. Some of the results are shown in *Figs. 2–5*. You can easily observe that, generally, modal (minimum pressure, deep green color) lines are not straight. They curve within the room and their locations are not immediately obvious without the FEM analysis. As I mentioned before, knowing your room nodal lines will help you determine where not to place your subwoofers.

Modal analysis revealed that the lowest modes are: F1 = 27Hz, F2 = 36Hz, F3 = 54Hz, F4 = 57Hz, F5 = 64Hz, and F6 = 73Hz. There are many more modes below 100Hz, but for the sake of clarity, I will focus only on those I mentioned. It is easy to observe that areas of maximum pressure (antinodes—deep red color for positive pressure or deep blue color for negative pressure) are always located near the walls or corners.

According to what I have just said, you would be tempted to locate your single subwoofer in one of those corners to take maximum advantage of the "room gain" effect. You may have also convinced yourself that you should place your subwoofer this way to allow the speaker to excite the maximum number of room modes, and preferably all room modes. This way, the smoothest (although still quite irregular) overall frequency response could be achieved. Here the "golden ratios" of room dimensions come into focus. The idea is to position room modes evenly across the low-end frequency range, which you can do by affecting room geometry.

MODAL ANALYSIS

Anyway, armed with all this common knowledge, I continue looking at *Fig. 2*, which, at first, you may find quite contradictory to what I just said. Why? Well, I have just said that pressure maximums are located at the room's corners. But, in *Fig. 2* you'll note corner "A," with the 26.32Hz nodal line (minimum pressure) sitting right at it. This is exactly the opposite of my previous statement . . . or is it?

Before I go any farther, let me point out the advantage of FEM employed for this analysis. Of course FEM is complex, but it allows you to see and model things that would not be readily possible without it. Now, back to solving the problem.

The answer lies in the room's "L-shaped" geometry. The corner "A" is located almost exactly halfway between corners "B" and "C." As you can see in *Fig. 2*, the 26.32Hz mode would develop between those two corners, so the pressure maximum would be located in those corners. It should be easy to envisage that the nodal line for this particular mode should fall right in the middle between these two corners. If you could "unfold" the room, (imagine corners "C," "A," and "B" lined up), you would find corner "A" sitting in the middle of the long wall marked by corners "C" and "B."

The problem actually becomes worse. If you examine *Fig. 5*, you will find the same issue at 68.22Hz. This is not exactly the harmonic of the 26.32Hz mode because the room has shorter walls on one side than the other. Once again, corner "A" has a nodal line right across it. You may expect this to happen at higher frequencies as well. I hope this short explanation of *Fig. 2* and *Fig. 5* offers you some indication of the importance of modal analysis, because it offers significant insight into the physics of your room acoustics.

ANALYSIS OF ROOM CONTRIBUTION

As a test case, I decided to place my subwoofers as described in *Fig. 1*. Now, having performed the modal analysis, I anticipate that there will be differences between subwoofer 1 and subwoofer 2 SPL plots. First, I need a reference "room contribution" coming from both subwoofers. This reference plot is shown in *Fig. 6*.

As I discussed before, there will be frequencies, where there are pressure minimums at the listener location at the wall. The most evident are 32Hz, 136Hz, and 184Hz, where the outward radiating pressure waves combined destructively and produced a node at the listener 1 location. The "notches" due to this are fortunately quite narrow.

You may notice that the frequency-response horizontal scale is linear, and not logarithmic, as it would be typically used. The "room contribution" is without a doubt quite irregular in comparison to a typical frequency response of a loudspeaker measured in an anechoic chamber. *Figure 6* represents a typical situation you may expect in your home. The room is an enclosed space and will resonate at its modal frequencies. I have selected a "Medium absorption" scenario for the purpose of this analysis, and this results in a gradual "smoothing" of the "room contribution" curve when moving towards higher frequencies.

If I used a lower absorption coefficient in my model, the "room contribution" curve would continue to exhibit sharp peaks and valleys within the whole frequency range of the analysis. I chose this type of approach for the purpose of better visualizing the pressure patterns, as the analysis progresses through the whole frequency range.

SINGLE SUBWOOFER CONTRIBUTIONS

The next task to perform in my analysis was to plot "room contribution" due to a single subwoofer. At the start, I decid-

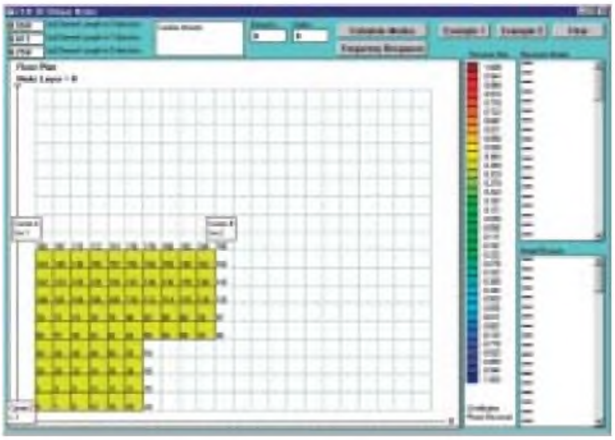

FIGURE 1: "L-shaped" listening room floor plan. B-2006-1

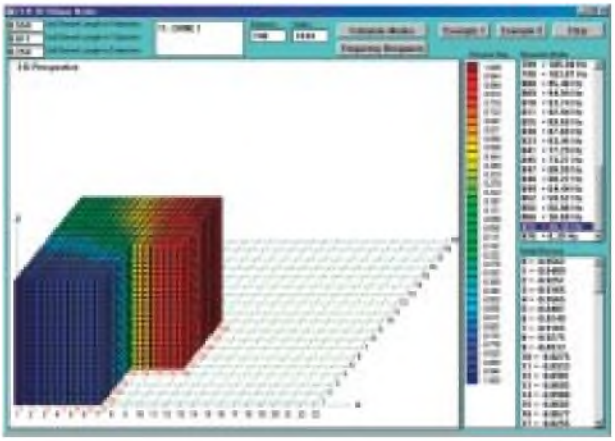

FIGURE 2: First mode at 26.32Hz. B-2006-2

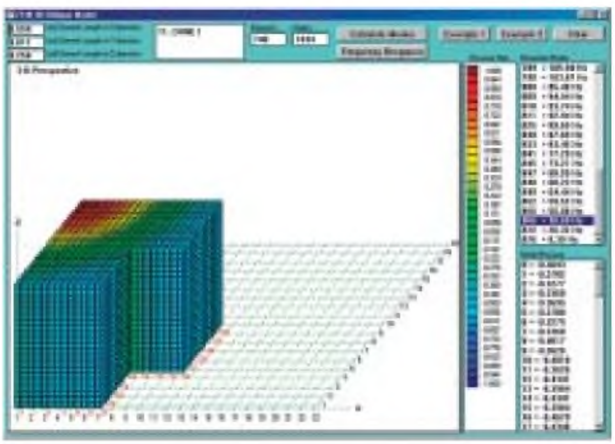

FIGURE 3: Second mode at 36.60Hz. B-2006-3

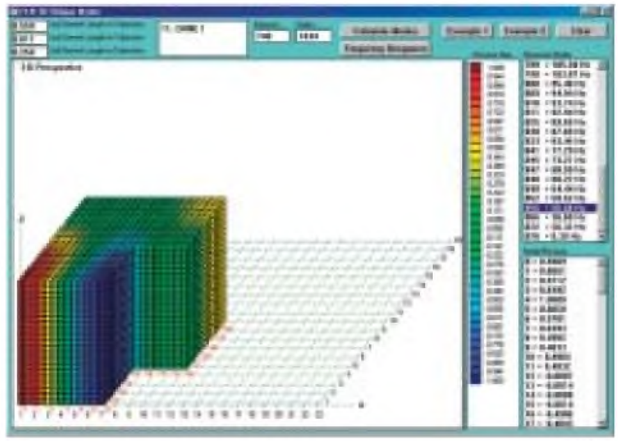

FIGURE 4: Example of mode at 56.08Hz. B-2006-4

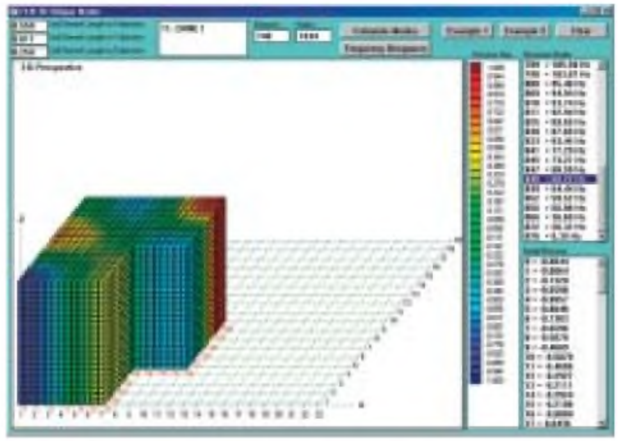

FIGURE 5: Harmonic mode 68.22Hz with nodal line in the corner. B-2006-5

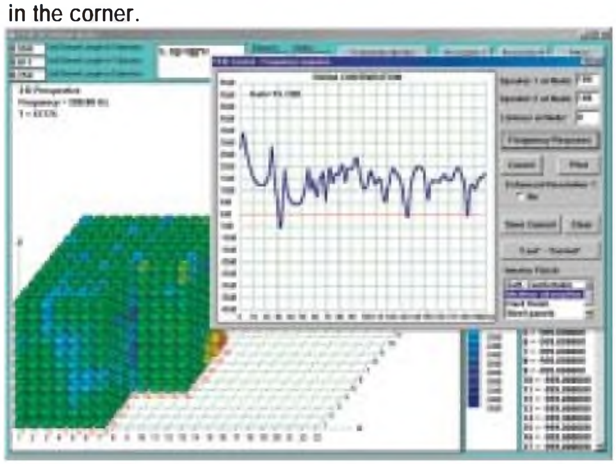

FIGURE 6: Reference "room contribution" coming from both subwoofers. B-2006-6

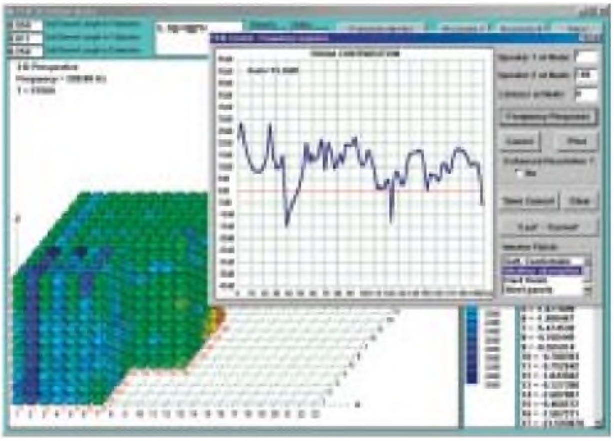

FIGURE 7: "Room contribution" due to single subwoofer Sw2 at node 186.

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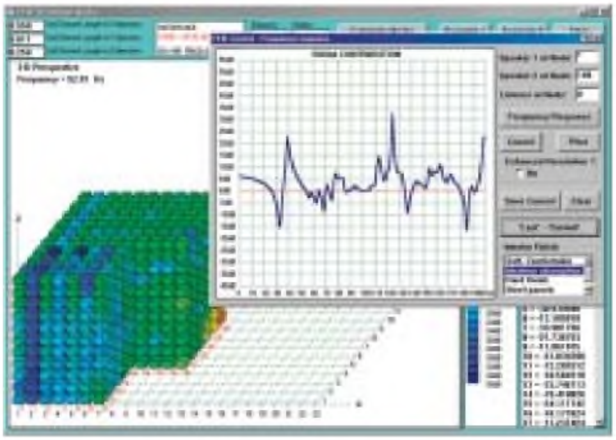

FIGURE 8: Difference between two subwoofers and a single (Sw2) subwoofer.

B-2006-8

ed to try subwoofer 2 (Sw2) only. This speaker is located in corner "B" at the node 186. I plotted the resulting "room contribution" in *Fig. 7*.

It is easy to notice that this curve is significantly more irregular than *Fig. 6* ("room contribution" due to both subwoofers). Keep this in mind when considering whether the number of subwoofers makes any positive difference. Frequency bands 35–50Hz, 110–140Hz, and 150–175Hz exhibit 8–12dB lower level.

Also, modes 26.32Hz and 56.08Hz are strongly present in this plot, which can be explained with the help of *Figs. 2* and *5*. In both instances the subwoofer (Sw2, node 186) and the listener (L1, node 0) were located at corresponding antinodes for these frequencies.

Finally, poor response in 35–50Hz frequency band is associated with the 36.60Hz mode. Pressure pattern for this mode is shown in *Fig. 3*, and pressure maximum is located at node 166, which is where the missing driver (Sw1) was located. The source is missing, so the mode will not be fully excited.

Another interesting plot is shown in *Fig. 8*, where the difference between two subwoofers versus a single (Sw2) is depicted. Everything that lies above the 0dB line indicates the advantage you are getting by using a dual subwoofer configuration, as opposed to only one. When you switch on the second subwoofer (Sw1), the total radiated power is only 3dB higher. However, inspecting the curve on *Fig. 8*, you may notice more than 3dB SPL gain in quite a few frequency bands: for instance, below 20Hz, 35–50Hz, 110–135Hz, and 155–180Hz.

The "dual-woofer advantage" approaches 6dB for the frequency range below the first mode (below 20Hz). This is quite correct, as the distance-related phase difference between the two woofers becomes smaller and smaller towards lower frequencies. The outputs from both woofers now add coherently (in-phase) and pressure simply doubles, resulting in a 6dB gain. This result is the same as if you put two woofers in a box twice as big and took advantage of mutual coupling between the woofers at the lowest frequencies. However, you may find it easier to deal with two smaller subwoofers rather than one box, twice as big[4].

The next step involves plotting a similar set of curves for subwoofer 1 (Sw1). *Figure 9* shows "room contribution" due to a single subwoofer–Sw1 at node 166.

Evidently, this SPL curve is poor and even more irregular than the result for single subwoofer Sw2. You will immediately notice the missing spectrum around 26Hz and 56Hz, as compared to dual-subwoofer operation.

I explained this problem when discussing the results of modal analysis. Now, you can finally see the effect of placing the loudspeaker on the nodal line on the overall SPL curve. It is evident from *Fig. 9* that subwoofer Sw1 fails to energize modes 26Hz, 56Hz, and so on. Even a simple visual inspection of *Fig. 9* is sufficient to see that corner "A" is not as good a location for a subwoofer as corner "B."

If you are the happy owner of a single-subwoofer system, you may need to do more homework on subwoofer placement than users of two-subwoofer systems. Corner "A" (node 166) may have been the choice of many users, simply because of its somewhat central location. Resulting "room contribution" would be poor, which is evident in *Fig. 9* and even more evident in *Fig. 10.*

One option would be to move the subwoofer out of the offending corner. *Figure 10* reveals the substantial contribution of subwoofer 2 (corner "B") to the overall SPL. This subwoofer dominates below 30Hz, 40–60Hz, 80–110Hz, and more.

CONCLUSIONS

Summarizing the analysis of my "L-shaped" room, I conclude that:

1. For the chosen example locations, each of the subwoofers alone will not produce SPL as good (level and smoothness) as two subwoofers played simultaneously in their respective locations.
2. For a single subwoofer, corner "B" is a better location than corner "A." Further modeling (recommended) is likely to reveal perhaps even better locations than "A" or "B."
3. Placing both subwoofers in corner "A" or in corner "B" will not result in a "smoother" SPL response; it will only raise the plots on *Fig. 7* or *Fig. 9*, respectively.

I have arrived at these conclusions working through my example, and I have accomplished the following:

1. I have determined modal frequencies and pressure patterns for all modes below 200Hz.
2. I have identified an "offending corner"–corner "A," where the subwoofer would miss some modes.
3. I also generated an SPL plot for both

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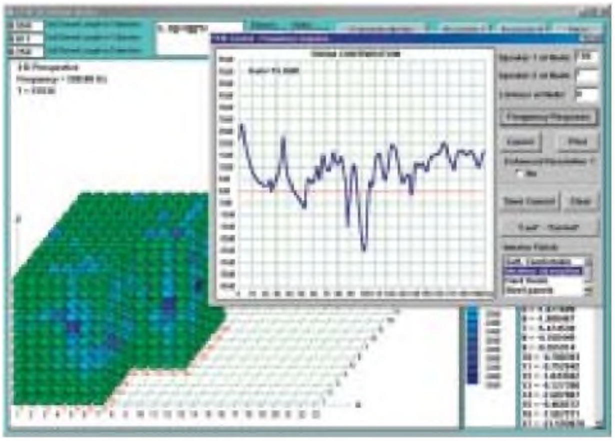

FIGURE 9: "Room contribution" due to single subwoofer Sw 1 at node 166.

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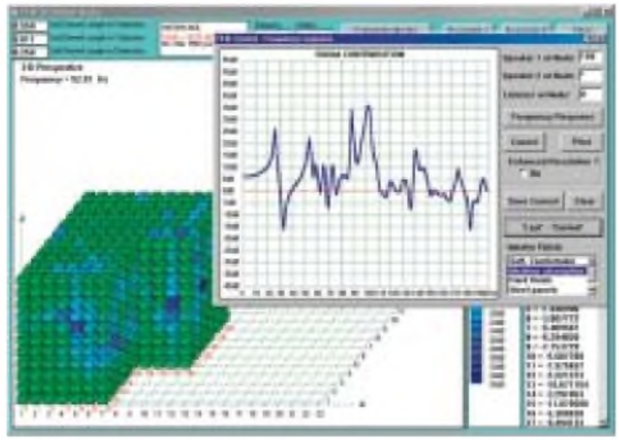

FIGURE 10: Difference between two subwoofers and a single (Sw 1) subwoofer.

B-2006-10

subwoofers and listener at chosen locations.

4. Then, I produced the SPL of individual subwoofers at their locations.

5. Finally, I plotted final curves showing the "dual woofer advantage" over the frequency range of interest.

The example I presented here does not attempt to justify these particular speaker locations. And by no means was the location of subwoofers considered to be optimal.

My goal was to better understand a multiwoofer setup and explain the application of the FEM, which is considered to be the most accurate modeling tool for this kind of analysis[2]. Particularly if the shape of your room cannot be handled by simple closed form equations. This would also be true for many of today's contemporary, open plan dwellings[6].

The FEM requires quite a bit of RAM and a lot of megahertz propelling your processor, so be prepared for lengthy analysis. My way around it was to set up the analysis on my Pentium III (500MHz) processor and then go and do my weekly grocery shopping. Two hours later, "room contribution" plots were ready. Now, do you still think that the speaker is the weakest link? ❖

Would You Take the Stereo?

Wildfires are once again ravaging the West. But a couple of years ago, several families in New Mexico faced the consequences of a wildfire. Fortunately, they survived the devastation. But how did their audio equipment fare? **By Nancy MacArthur**

On May 11, 2000, we left home abruptly in the middle of the night as a wildfire approached our town. Although we took a few possessions, we didn't take the stereo. We've wondered why ever since.

I'm not implying that audio equipment isn't important to us. We love music and listen to it constantly. In the years when we had little money, we spent an enormous fraction of our tiny income on audio equipment. We've spent years putting together a system we enjoy, one piece at a time. But when push came to shove in a crisis, we left the stereo behind.

LEAVING LOS ALAMOS

Duncan and I started worrying about the Cerro Grande fire on Saturday. From a hill near our house in White Rock, we could see flames leaping on the mountainside as they approached Los Alamos, ten miles away. We wondered whether friends in Los Alamos would have to leave and if anyone would come stay with us. On Sunday a small-scale evacuation of the Los Alamos neighborhood nearest the fire began.

By Monday the schools and most workplaces in Los Alamos and White Rock had shut down. We cleaned the house and stocked up on groceries.

The fire lines held on Monday and Tuesday. On Wednesday high winds blew firebrands over the lines, and the entire population of Los Alamos–men, women, children, babies, old people from the nursing home, pets–evacuated the town.

By midafternoon three families and seven extra cats had joined us. Kathy and Rick arrived first. Kathy shook her head as she came through the door. "Our house may not make it," she said. "We're only a block from the forest."

Linda came next with her eleven-year-old daughter, Laura. Kathy's parents, Martin and Judy, drove out on an old dirt road opened temporarily for the evacuation. They arrived last.

Shortly after the evacuees left, the fire leaped into Los Alamos. Firefighters later described a fast-moving, hundred-foot-high wall of flame with a roar like a jet engine. As they fought the flames, embers the size of baseballs flew over their heads.

In our family room in White Rock, seven adults and two children crowded around the television. Four traumatized cats huddled together in a single cat carrier. The children's eyes grew big as they watched Los Alamos start to burn.

Linda's sister in Albuquerque had invited Linda and Laura to stay. They drove to Albuquerque and moved in with her. Old friends living nearby offered Martin and Judy a spare bedroom. They moved on. Rick and Kathy and four cats remained with us.

We fed Kathy and Rick a good dinner and reassured them that the fire was still ten miles away. The news coverage of the fire throughout the evening remained grim.

We tried to go to bed early, but nobody slept well. The wind howled outside the windows.

EVALUATION

At 2:00 AM when the phone rang, I'd been asleep only a short while. I stumbled to the phone. The caller was Martin, Kathy's father, who was staying with friends nearby. He said the evacuation of White Rock had begun and we should prepare to leave.

I stared blearily at the phone. Los Alamos, I thought, they evacuated Los Alamos a few hours ago. Martin repeated that the evacuation order was for White Rock.

Aerial photo from news helicopters confirmed many residents' worst fear—total destruction of their homes. Entire neighborhoods were devastated during the night. (From the book, *Cerro Grande: Canyons of Fire, Spirit of Community*)

ABOUT THE AUTHOR

Nancy MacArthur has written for *Speaker Builder, Audio Amateur, Glass Audio*, and *audioXpress*. She has lived in the American Midwest and on the East Coast, as well as in Costa Rica, Colombia, and the U.K. Currently Nancy, her husband Duncan, and their young son Colin have settled in the high desert of northern New Mexico.

"Okay," I said, "We'll meet at my father's house in Santa Fe." "What's the address?" asked Martin. I drew a complete blank. "I can't remember!" I wailed.

During the night high winds had whipped the wildfire down Pajarito Canyon toward White Rock. Our town was no longer safe.

Duncan and I ran through the house throwing clothes, photograph albums, important papers, and some small family heirlooms into suitcases and pillowcases. Rick and Kathy, by now experienced refugees, helped us pack.

Colin, our ten-year-old, got up and packed his backup computer disks, his old teddy bear, and two books. He climbed to a high shelf and rescued his father's ancient toy leopard. I woke up enough to remember my parents' address and drew maps of the route for Kathy, Rick, Martin, and Judy. I called friends to make sure they'd heard about the evacuation. In the chaos nobody remembered the audio equipment.

Although the streets were clogged with traffic, Martin and Judy managed to drive to our house on the back roads. We threw a few possessions into the cars and started to caravan out of town.

THE WAY TO SANTA FE

Only one road out of White Rock remained open since the fire had cut the other road days before. In addition, the population of the town had swelled, as thousands of Los Alamos evacuees moved in with friends in White Rock. The result was gridlock. As the northwest sky glowed orange and the local radio station urged us to leave town immediately, the lines of cars remained motionless.

Two hours later the cars started moving. We reached Santa Fe at dawn.

Towns throughout New Mexico struggled to find shelter for 18,000 suddenly homeless people. Duncan, Colin, and I, along with my sister and her family, crowded into my parents' house in Santa Fe. Kathy, Rick, Martin, and Judy moved in with kind strangers.

Local television stations provided continuous coverage of the fire. Sometimes we'd see reporters standing in front of an intact city block, and we'd rejoice that a friend's house had survived. Other images were less encouraging. As a house with an unusual peaked roof went up in flames, we watched in shock. "That looks like Rick and Kathy's house," I whispered to Duncan. "I hope you're wrong," he said.

News from White Rock was sketchy and contradictory. The fire had jumped State Route 4 near White Rock. No, the fire was still three miles away.

At this point we remembered the stereo system we'd put together with effort and expense over the years. We mentally listed the components—some built by hand, others purchased new. Each piece had gone through a process of careful auditioning and testing before its addition to the system. We remembered our collection of Mobile Fidelity gold CDs, some nearly impossible to replace. We thought guiltily of the Assemblage SET-300B sent to us for review, still sitting in our house in White Rock with the other audio equipment.

Returning to White Rock to retrieve the equipment was not feasible. The National Guard had set up roadblocks and took a dim view of local residents' attempts to cross the barriers.

PHIL

Later I discovered that other audiophiles had thought more carefully about what to do with their equipment during the emergency. Phil, for example, knew his house in Los Alamos was at extra risk due to its location on a canyon. (Wooded canyons sometimes act as conduits for wildfires.)

As ashes from the approaching fire fell on his deck, Phil considered his audio system. Taking his speakers was out of the question, as they weighed 180 pounds each. The turntable was delicate; it might be safer staying at home. The CD player meant less to him than the turntable; it could stay behind. What about the electronics? Phil had built his preamps and amplifiers from old *Audio Amateur* designs. Perhaps if the house burned he could obtain copies of the original designs and remember the modifications he'd made.

Deciding which recordings to take was more difficult. Taking them all was not feasible, as Phil owned over 5,000 LPs and a slightly smaller number of CDs. Which to choose? Perhaps his collection of Mercury LPs, out of print and difficult to replace? On the other hand, audiophile recordings aren't necessarily what a music lover would save. Should he take Beethoven or Shostakovich string quartets? Verdi or Mozart operas? Impossible to decide.

Phil had entered all his recordings on a Microsoft Access database, and he had receipts and photos of his audio equipment. He decided to leave equipment and recordings behind and assume that insurance would partially replace them. He and his wife Barbara took the dog, a few personal items, and their collection of oriental and Navajo rugs.

ANDY

Andy didn't worry much about the oncoming wildfire. Fifteen years earlier his apartment in Texas had caught fire, and he knew insurance would replace many of his possessions.

Andy also didn't think the fire would reach his house. The Los Alamos Medical Center lay between his house and the fire, and he assumed the firefighters would make a stand in front of the town's only hospital.

On Wednesday he watched the smoke from the fire grow thick. When the evacuation order came, Andy left behind two laser turntables, a roomful of high-end audio and video equipment, and 5,000 LPs, including rare recordings: a Southern Heritage collection of folk music as well as experimental stereo recordings from Bell Labs in the 1930s. He also left German records from 1944 of Beethoven's *Ninth* and Tchaikovsky's *1812 Overture*, which had recorded background sounds of antiaircraft fire and Allied bombs.

Some of Andy's equipment and recordings were fragile. "That stuff is safer at home than on the road," he said. "Unless you know for sure that your house is going to burn."

When the order came, Andy thought the evacuation would last only a few hours. He took along a student who had been working with him and an overnight bag.

Months later Andy spoke of the information overload that accompanies a crisis situation. "The information input was hard to deal with," he said. "You have to go with the best information you have at the moment. And you can't always count on behaving rationally. During World War II my aunt left Poland just ahead of the Russians. All she took with her was a closetful of ties."

P.H.

P. H. thought his house near the center of Los Alamos probably wouldn't burn; he nonetheless made preparations in the days leading up to the fire. He drove his '48 Jaguar drop-head coupe and his '37 Bentley to White Rock and left them there. On Monday he packed his other cars with photographs, family memorabilia, paintings, and manuals for the old cars.

He didn't pack his Mark Levinson amplifier, KEF 104 reference speakers, and other audio equipment. He considered taking equipment but decided it could be replaced.

When the wind shifted away from Los Alamos on Monday and Tuesday, P. H. and his wife Marilyn unpacked the cars. On Wednesday when the evacuation order came, they had to repack quickly. They took the irreplaceable items mentioned earlier, along with their poodle and three pet skunks.

P. H. later regretted not packing his cars full to the rooftops. "I could have taken more car parts," he said, "And my CD collection."

THE OLSHERS

Dick Olsher, senior editor of enjoythemusic.com, and his wife Lesley were at home in White Rock watching the fire news on Wednesday night. When the power went out in Los Alamos around 11:00 PM, cable television in White Rock failed. Dick rigged rabbit ears from a coat hanger and tuned into a local station.

The broadcast of the evacuation order for White Rock at 1:30 AM came as a shock. The Olshers, like many White Rock residents, had been thinking about Los Alamos. Few White Rock people had expected to flee their homes that night.

An AP reporter knocked on their door and asked if she could observe them preparing to leave. As she watched, Dick and Lesley packed pictures, tax records, and other important papers. They took the master tape for Lesley's CD *Jazz Me*, recorded with David Manley in 1992. They took the dog and the birds; the fish in their tank would have to stay behind.

Dick, a long-time audio reviewer, had plenty of stereo equipment. He didn't take it. "I wouldn't jump into a fire to retrieve a piece of audio equipment," he said later. "Audio is important, but it's not as important as family, as putting together the things that make you who you are and what you are."

He added that in some cases incinerating audio equipment could be a positive experience. "Some audiophiles might feel better if they put their equipment on a bonfire," he said. "Especially when they've made a mistake."

Meanwhile, back in Santa Fe the evacuation dragged on. Local television stations replaced footage of burning houses with footage of ashes blowing over foundations. Portions of Los Alamos National Laboratory went up in flames. The fire moved down Mortandad Canyon toward White Rock, then spread north onto the Santa Clara Indian reservation.

Duncan, Colin, and I searched for missing friends. Nearly everyone we knew had scattered to parts unknown in the middle of the night. We found them one at a time by chance, by word of mouth, and over the Internet.

On Friday another pair of kind strangers told us they were leaving town for a week and offered us the key to their house. We moved in temporarily.

AFTERMATH

The evacuation order for White Rock was lifted late Sunday afternoon. On Monday morning we returned home to a house that was smoky but still in existence. Our audio equipment was intact.

The audiophiles mentioned earlier found themselves in various circumstances as they came home. Phil, P. H., and Dick had houses untouched by the fire. Their equipment was undamaged.

Andy, who had planned to be absent only a few hours, had to wait over a month to move back into his house. Although the fire never reached Andy's neighborhood, his house had suffered extensive smoke and water damage. Water from his pipes had flooded the floors, possibly due to changes in pressure during the firefighting.

Andy's audio equipment, which had been kept up off the floor, was undamaged. His records had been stored in their sleeves above the floor. They were also intact.

The people who were with us during the evacuation came back to diverse situations. Linda and Laura's house survived, although Laura awakened two weeks after they returned home to find a resurgent fire burning in the canyon below their house. A last-minute shift in the wind had saved Martin and Judy's neighborhood.

Kathy and Rick's house burned to the ground, along with the homes of 353 other families. Their audio equipment is gone, as well as the grand piano and the Thai antiques an uncle had left them. They have the few possessions they brought to our house that Wednesday afternoon, along with a handful of items (pottery, mostly) that survived the flames.

REFLECTIONS

Since the fire we've asked ourselves what we've learned from the experience. Would we make different decisions another time? Of course. Next time we'd be more aware of a wildfire's ability to move far and fast. We now keep a room-by-room list of items to take with us in an emergency. Our rare CDs are shelved separately from the rest, so we can throw them in a pillowcase and leave quickly.

Would we take the stereo in the future? It depends. If we had plenty of warning, enough time to pack fragile tube amps and heavy speakers, yes. In another middle-of-the-night, run-for-your-lives situation, no. In that case, I would still choose to use what little time I had to warn friends and save possessions of sentimental value.

In an emergency each individual will have different priorities and make different choices. As I see it, we could replace our audio equipment, albeit with heartbreak, effort, and expense. We could not replace friends and keepsakes. Another time I would first call friends and pack photographs and small family heirlooms. Then, if time and space remained, I'd take the stereo.

Or maybe I wouldn't. Dick Olsher summed up the reality of an emergency situation well. "You think you know how you'll respond in a crisis," he said. "But you may not be thinking straight when it happens."

Dick has a point. I would suggest thinking about which belongings have

(to page 69)

Product Review

EarMax Pro Tube Headphone Amplifier

Reviewed by Charles Hansen

Audio Advancements LLC, PO Box 2090, Branchville, NJ 07826, 973-875-8705, www.audioadvancements.com, e-mail audadv@earthlink.net, $750 U.S. Dimensions (EarMax): 3.75″ W × 3.5″ D × 4″ H (top of tube protectors); 15 oz (Power Supply): 2.13″ W × 3.5″ D × 2.25″ H; 15 oz (ABS), 10 oz (wood). Limited two-year warranty (tubes one year).

The EarMax Pro is a single-ended tube headphone amplifier rated for 150mWpc (per channel) into headphones with impedances from 35Ω to 1k. I used Grado SR-125 headphones (32Ω) to check for proper operation of the EarMax Pro. The standard EarMax ($575 U.S.) is rated for less output power (100mWpc), and requires headphones of 200Ω to 2k.

The standard unit is made of cast ABS plastic. A custom model with a handcrafted, solid wood enclosure is also available. Optional accessories include a PakMax 12V DC rechargeable battery, DC-AC converter, cable, battery charger, and an optional carrying case for headphones, CDs, and CD player.

CONSTRUCTION

The EarMax is shipped in a custom two-compartment styrofoam block. Audio Advancements claims that the EarMax is the world's smallest vacuum headphone amplifier, and I see no reason to doubt it. There is a lot of circuitry packed into this tiny package. *Photo 1* shows the power-supply block (left) and the amplifier unit (right).

The power supply uses a three-pin cable with a Lemo connector that plugs into a mating jack at the rear of the amplifier. A rocker power switch and blue LED are located on the power supply. The brass volume control knob is located on the top of the amplifier, just forward of the three tubes. Gold-plated Tiffany-style RCA input jacks and a gold-plated ¼″ headphone jack are located on the sides. A brass guard formed into three loops protects the tubes from damage.

Photo 2 shows the amplifier with the aluminum bottom cover removed. All the circuitry is mounted on one double-sided epoxy PC board with solder mask. The only discrete wiring is from the input and power jacks to the circuit board.

Workmanship is first-rate. Wima film capacitors (some 5%) and Philips electrolytics are evident. The resistors are all 1% metal-film types, and the volume control is a dual 100k Alps Black Beauty. The tubes supplied with the EarMax were one ECC81 (12AT7) and two unmarked ECC88s (6DJ8).

TUBE-POLOGY

A schematic diagram was not supplied with the unit, but its layout is pretty straightforward. The EarMax circuitry is OTL (output transformerless) Class-A with low enough output impedance to drive the headphones.

The selected input signal is applied to the Alps audio pot. The wiper for each channel is connected to the grid of half the 12AT7/ECC81 dual triode. The plate of the first stage is coupled to the grid of a 6DJ8/ECC88 dual triode connected in shunt regulated push-pull (SRPP). The audio output signal is coupled to the output jack through a large electrolytic capacitor loaded with an output resistor. (The standard EarMax uses a lower-powered 6GM8/ECC86 output tube.)

The three tube heaters are connected in series across the 19V AC power-supply input, with the ECC81 filaments paralleled for 6.3V AC operation. This AC

PHOTO 1: Power supply and amplifier.

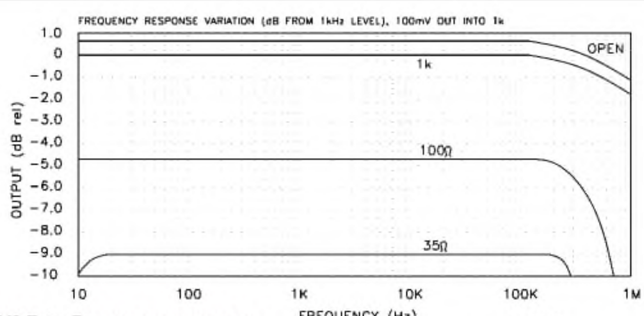

FIGURE 1: Frequency response.

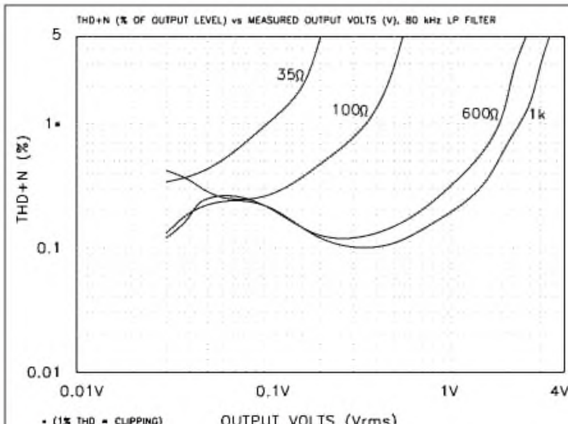

FIGURE 2: THD+N vs. output.

supply is voltage-doubled with diodes and large reservoir capacitors. The main reservoir caps are rated for 40V DC, so the tubes are running at fairly low B+ voltage, probably about +55V DC, since a voltage-doubler circuit produces a DC voltage of approximately twice the peak AC input.

MEASUREMENTS

I operated both channels of the EarMax Pro with pink noise at 150mV RMS into 35Ω (0.64mW) for one hour. The unit runs very cool. The input impedance measured 92k regardless of the volume control setting. The output impedances for both channels measured a low 85Ω at 1kHz (most mid-fi receivers use a resistor of 150Ω to 1k in series with the power amplifier output to provide the signal to the headphone jack). The EarMax preserves normal polarity.

The EarMax showed unity gain into 35Ω when the volume control was set at approximately 4 o'clock, and a 1k load reduced the unity gain setting to 1 o'clock. There was neither hum nor hiss in the headphones, regardless of the volume control setting, and it was absolutely quiet during power-up and shutdown.

I recorded the frequency response for loads of 35Ω, 100Ω, 1kΩ, and an open circuit (*Fig. 1*). The output coupling cap rolls the low frequency off –3dB at $f=1/(2\pi RC)$. The high end of the EarMax frequency response is specified to be 1MHz. In order to measure that lofty level, I needed to pull out two pieces of high-frequency equipment I don't use very often: an HP-204C oscillator and an HP-200D VTVM. High-frequency response rolled off gradually above 120kHz, with no additional HF gain peaking.

The EarMax provides a maximum gain of only 4.3dB with a 35Ω load. Full-volume gain increased to 14.2dB with 1k. Volume control tracking was excellent, with no measurable difference for output voltages from 30mV to 2.5V RMS into a 1k load. Channel separation (crosstalk) was a bit better from right to left (*Table 1*).

The EarMax's square-wave response was very good. The 1kHz response was just about perfect. The 40Hz square wave showed some tilt, which increased as the load decreased, as a result of the low-frequency response rolloff caused by the output coupling capacitor. The 10kHz response showed a very slight leading edge rounding with a 35Ω load, and one small damped cycle of peaking with a 1k load.

As with most tube gear, the EarMax never really goes into hard "brick wall" clipping. The positive peaks of the waveform are the first to be com-

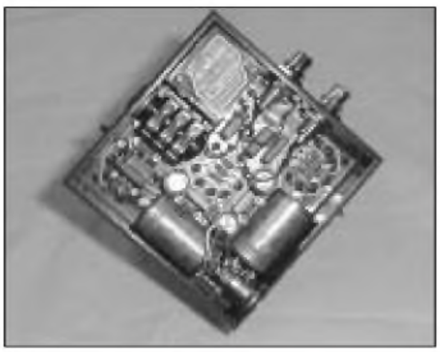

PHOTO 2: Interior view.

TABLE 1
CHANNEL SEPARATION

FREQUENCY (HZ)	R-L	L-R (DB)
100	–62	–58
1,000	–61	–57
10,000	–51	–43
20,000	–46	–37

TABLE 2
MEASURED PERFORMANCE

PARAMETER	MANUFACTURER'S RATING	MEASURED RESULTS
Bandwidth	4Hz–1MHz, ±3dB	6Hz–530kHz; +0/–3dB 35Ω 2Hz–1.1MHz; +0/–3dB 1kΩ
Power output	150mWpc	0.32mW, 35Ω, 1% THD+N 1.2mW, 100Ω, 1% THD+N 6.3mW, 600Ω, 1% THD+N 7.1mW, 1kΩ, 1% THD+N
Load impedance	35Ω to 1kΩ	
Output impedance	N/A	85Ω at 1kHz
Distortion THD+N	N/A	See *Fig. 2*
CCIF intermodulation	N/A	0.32% 35Ω, 0.283Vpp 0.14% 1kΩ, 0.283Vpp
Channel separation, 1kH	N/A	See *Table 1*
Hum/noise, input shorted	N/A	0.3mV RMS

PRODUCT REVIEW
AUDIO ADVANCEMENTS' EARMAX PRO HEADPHONE AMPLIFIER

LISTENING SETUP

I used the EarMax Pro in my listening room on a dedicated AC line and earth ground, fed by my home-built isolation transformer power conditioner and an AudioPrism Foundation III. The source was a Rotel RDD-980 CD transport paired with the Assemblage DAC-3 D/A processor. Headphones included the Sony MDR-CD999, Grado SR-80, and Grado SR-225. On hand for comparison were the Parasound ZAMP (used as a headphone amp), NAD 1300 preamplifier, and the Amphony Model 1000 digital wireless headphones.

Digital interconnect was Sound & Vision Digiflex Plus BNC; analog was either Kimber Silver Streak or Kimber PBJ. All components inhabited a RoomTune JustaRack, with the RDD-980 on three small AudioPrism Isobearings.

To ensure break-in, I let the EarMax play at fairly loud volume for over 50 hours before listening. A half-hour warmup every listening session brought the tubes to a good operating state.

REVIEWED BY MUSE KASTANOVICH

For frequency balance, any combination of headphone-amp will actually depend more on the headphones than the amp, although different amps, of course, have their own subjective colorations as well. First I tried my ten-year-old Sony MDR-CD999 headphones with the EarMax Pro (hereafter referred to as EarMax), with good results. On the *Buena Vista Social Club* CD (World Circuit/Nonesuch 79478-2), the Sony had a mellow sound that gave voices and drums a nice genuine feel, but guitars and rasp were a little too soft. This particular combination had a wonderful presence and lively dynamics that kept the music very involving.

Over the long term, however, the MDR-CD999 sounded just a bit too rolled off in the treble with the EarMax. Using the Silver Streak interconnects helped a bit in the top octave, but the frequency balance was still not quite right. In Sony's defense, I tried them briefly with the other amps, and the frequencies were a bit more balanced, but I ruled them out as the best match for the EarMax. Talk about extra volume headroom, these Sony headphones are so sensitive that I could not turn up the volume more than one quarter from zero on the EarMax.

Next up were the Grado SR80 headphones. Like the Sony unit, these have a screw-on ¼″ phone plug adapter, the native connector being a miniature ⅛″ stereo plug. I assume this is to make them easier to use with portable devices, but it's not the best thing for the signal to pass through one more mechanical/electrical contact.

The sound with these Grados was stunningly good, almost a match made in heaven. For example, on the recently remastered *Kind Of Blue* CD (Columbia/Legacy CK64935), I could feel the size of the bass' soundboard, see the shimmer flying off the cymbals, and catch every nuance of Miles Davis' musings. Nothing seemed blown out of proportion, and nothing was so small that it disappeared into the mix either. I used the Kimber PBJ interconnects.

With the Silver Streak, the top octave was a touch too hot. I recommend sticking with copper interconnects for this combination.

Lastly, I plugged in the Grado SR225 headphones, which sell for around $200, approximately twice the price of the SR-80s, and about what the MDR-CD999s originally sold for. These have a far more detailed and refined sound than the MDR-CD999s, however. They sound fairly similar to the SR-80s, but with a touch more detail and a smoother, more subdued, bass region.

ABOUT THE AUTHOR

Muse Kastanovich received a bachelor's in physics from Oberlin College. He was a contributing editor with *Stereophile* magazine for three years. He has enjoyed modifying all sorts of audio equipment. He has built two different versions of the Pass Zen amplifier and the Bride of Zen preamp. He is currently employed as a computer tech with a small shop in Albuquerque, N. Mex.

The SR-225s are also a bit more dynamic and articulate. They spacially separate the different instruments more, and are probably the best-performing headphones I've ever heard in that department. Both pairs of Grados share an uncanny (no pun intended) ability to place instruments not just inside your head, but arranged just outside it as well, and this pleasing quality was greatly facilitated by the EarMax. Instruments were usefully separated in space, not as much as with well-placed loudspeakers, but enough to help me notice every single musical line in a composition.

So for the rest of the review I used the most flattering headphones for the EarMax—both pairs of Grados. You might argue that this is not fair in the comparisons that follow, because the frequency balance with the MDR-CD999 was actually better with some of the other amps. But there is much more to judging quality than just frequency response, and the Grados were superior in most other areas of presentation, regardless of amp.

Not surprisingly, Audio Advancements' literature on the EarMax Pro states that they were designed to drive low impedance headphones such as the 32Ω Grados. From a listening standpoint, they seem to have succeeded in that goal.

DIFFERENT AMPLIFIER COMPARISONS

First, I compared the EarMax to the interesting Amphony model 1000 digital headphones with their own built-in headphone amp. At $129 retail, the whole thing sells for about one-seventh the price of the EarMax/Grado combination! I did not expect such an economical choice to really compete with the EarMax sonically, and on the finer points of performance it did not. However, these wireless phones designed in Germany sounded quite a bit better than I had expected for the price.

The Amphony had dynamics that were nearly the equal of the EarMax, and had quite a pleasing, fairly accurate frequency balance. In other areas, such as transparency, spaciousness, pace and rhythm, and timbre, the EarMax was preferable though. The EarMax had a smoothness and a naturalness to it that went a long way towards helping propagate the illusion I was listening to something real. The Amphony had more of these pleasing qualities than I had expected, though it really could not compete with the EarMax in overall fidelity and involvement.

Next, I compared the EarMax with a ten-year-old NAD 1300 preamp. Here the NAD serves well as a reference to the average sound quality that you might expect to get from a receiver, integrated amp, or preamp that has been hanging around the house for a few years. The NAD tended to mush different instruments together into a slightly amorphous mass of sound, as compared to the EarMax, which delineated them well from one another.

The EarMax was also more dynamic, with quicker-sounding and more dramatic drums and bass guitar transients. In terms of individual instruments, the EarMax gave each a nice coherence and realistic body, as opposed to the NAD, which gave them some rough graininess that took away from the realism. The NAD had a thinness to its sound, not just in terms of frequency response, but also in the decay of each note and the reverberation.

The EarMax was sonically superior to the NAD in just about every department, except perhaps the top octave, where the NAD had just a touch more of the pleasing shimmer. The EarMax was superior in rhythm, dynamics, bass weight, detail, soundstaging, and definitely the kind of mellowness that you might take for granted at a live performance. The EarMax was much more involving overall, but in its defense, the NAD sold for less than the EarMax when it was new, and most of the resources were dedicated to other parts of the preamp than the headphone output.

THE PARASOUND ZAMP

The Parasound ZAMP (Zone AMPlifier) is a small (half chassis size) power amplifier rated at 30W per channel, with a headphone jack that is in parallel with the speaker outputs. It serves well as a dedicated headphone amplifier, particularly for those who like to listen loud, with plenty of power in reserve and a nice low output impedance. It is much less expensive than the EarMax, so think of it not as a direct competitor, but rather as a sonic reference point the EarMax should be able to surpass to be worth a recommendation.

The Zamp did not have the raggedness in the treble that plagued the NAD's reproduction. It could not match the inner glow of the EarMax though. I apologize for using such a nondescript audiophile phrase, but that is just how the EarMax makes you feel. It imparts the music with a life and immediacy that bring it closer to sounding real than the other amps.

Reverberation—whether real or artificial—was more pleasing through the EarMax. It was more liquid, more enveloping. The EarMax gave individual instruments and voices more coherence as well. The Zamp's dynamics were very good, but not quite the equal of the EarMax's. Its frequency balance was almost identical to the EarMax, though, which helped in hearing the other differences.

Unlike the NAD, the ZAMP had the full measure of bass weight that the music demanded. It came much closer to the sonic performance of the EarMax in other areas as well, but could not quite equal it. It fell short in terms of absolute transparency. Small details were easier to hear through the EarMax.

Nor could it match the smoothness and the extraordinary finesse with which each note decayed. Individual instruments had a solidity which was almost touchable with the EarMax. The sense of space was superior with the EarMax as well. I can't remember ever feeling so surrounded by a band or orchestra through headphones as I did with the EarMax.

CONCLUSION

What first caught my attention listening to the EarMax was an overall balanced sound. Some say this is the most important quality any component can have, though there certainly are other elements important to true high-end reproduction. I don't mean just a balanced frequency response, but a balance between various different strengths that gave CDs a natural realism that allowed listening right down into the music. The EarMax Pro maintained this impression, and deepened it, all the way through to the end of the review period.

The EarMax Pro is an extraordinary-sounding headphone amplifier. It might be a bit pricey for such a device, but for someone who listens through the cans a lot it's worth every penny. Apartment dwellers, loud-volume fanatics, and those in crowded houses could be quite happy with just a headphone system built around this amp.

It is very involving and maybe even addictive. I found myself needing to dance every time I clamped those Grados on, which tells you something.

It's not perfect, but it has no major sonic flaws, and I highly recommend it.

pressed with increasing output voltage. The left channel distortion was higher than the right (*Table 2*). As you can see, there are not very many published specifications with the EarMax.

The maximum open-circuit output voltage swing was +5, −6V pk (3.9V RMS on my true-RMS meter), decreasing to a maximum of 0.25V RMS at 35Ω. This made it impossible to achieve the rated 150mW per channel, regardless of the load (*Table 2*). The absolute maximum output power (>10%THD) measured 1.26mW for 35Ω, 3mW for 100Ω, 10.4mW for 600Ω, and 15mW for 1k. I think that the specification sheet has a misprint, and the output should read 150mV. I found an uncomfortably loud sound level just 100mV into my 32Ω Grado headphones, where the distortion measured 0.95%.

Figure 2 shows THD+N vs. output voltage into four different loads at 1kHz. I engaged the test-set 80kHz low-pass filter to limit the out-of-band noise. The 1% THD+N clipping voltage measured 105mV RMS for 35Ω, 345mV RMS for 100Ω, 1.75V RMS for 600Ω, and 2.47V RMS for 1k.

The THD+N at a fixed output level does not vary significantly over the audio-frequency range. The biggest change (1.2% at 20Hz down to 0.95% at 63Hz) occurs with the 35Ω load. Above 63Hz the THD vs. frequency for all loads is flat to 20kHz.

The residual distortion waveform for 2.4V RMS into 1k (7.1mW) at 1kHz is shown in *Fig. 3*. The upper waveform is the amplifier output signal, and the lower waveform is the monitor output (after the THD test-set notch filter), not to scale. This distortion residual signal consists mainly of the third harmonic riding on a 120Hz power-supply component. THD+N at this point is 1%.

A repeat of this test with 100mV RMS into a 35Ω load (0.3mW) is shown in *Fig. 4*. Here, some high-frequency noise is visible on the third harmonic. THD+N at this point is also 1%.

The spectrum of a 50Hz sine wave at 100mV RMS into 1k is shown in *Fig. 5*, from zero to 1.3kHz. The 50Hz fundamental at −18dB is my 0dBfs reference point. The THD+N measures 0.22%, with the second harmonic measuring −72dBfs and the third −85dBfs. The −54dBfs 120Hz power-supply component makes up a significant portion of the THD+N, with further components at 180Hz, 420Hz, 480Hz, and 840Hz. The few additional 50Hz harmonics are all below −95dBfs. When I removed the power-supply harmonics from the THD+N computation, using only the actual harmonics of 50Hz, the THD dropped to 0.026%.

I increased the 50Hz signal level into 1k to 2.4V RMS (not shown), and the THD+N measured 1%, with the second harmonic at −50dBfs and the third at −41dBfs. The power-supply components were now much lower in the spectrum, below −85dBfs, adding only 0.018% to the total THD+N. The hum/noise level with the inputs shorted was 0.3mV RMS, regardless of the volume control setting. Hum and noise may be more audible with higher sensitivity headphones.

A repeat of the 50Hz, 100mV RMS spectrum with a 35Ω load is shown in *Fig. 6*. The second and third harmonic are just about equal at −43dBfs, but the power-supply components are all below −70dBfs. THD+N measures 1.02%. A spectrum analysis of a 1kHz fundamental (not shown) shows the third harmonic to be dominant. This suggests that the slight increase in THD below 63Hz is due to a higher second harmonic.

The CCIF intermodulation distortion (19+20kHz) at 0.283V p-p into 35Ω was a high 0.32%. As I increased the load to 100Ω, then 1kΩ, the IMD at 0.283V p-p decreased to 0.26% and 0.14%, respectively. The CCIF IMD graphs for 1k and 35Ω are shown in *Fig. 7* and *Fig. 8*, respectively. The nonlinear tube transfer characteristics produce a wide range of intermodulation products.

A 0.283V p-p multi-tone IMD signal (9kHz + 10.05kHz + 20kHz) produced a very high 1kHz IMD product of 5% into 35Ω, and 0.25% into 1k. ❖

Manufacturer's Response:

Thank you very much for the extremely careful and precise way you measured this little unit. No one—except maybe myself—found this necessary before, despite all the reviews we've had.

I'm sorry for not supplying a schematic with the unit, but I did not know about the plans for a review before it was shipped. On the other hand, I want the Far East and other "We copy everything" people to use their own ears and brains or to buy one unit.

But as expected from a designer of your class, you managed easily to describe how Ear-Max Pro works without the schematic. So there are only a very few details to be corrected.

A. The bottom is not made of aluminum but of stainless (antimagnetic) steel.

B. I must add to your description of the voltage doubling for the plates' B+ voltage. The 19V AC input for the tubes filament is doubled by an auxiliary voltage (think of the third pin of the Lemo-connector) before it is given to the voltage doubling with the diodes and reservoir caps. This results in a plate-supply voltage of ~80V DC.

Because of the relatively high internal power-supply impedance and the current drawn by the tubes, the peak voltage of the doubler is not reached. The plate voltage is still very low (<40V), securing a long tube life and a longer life for our ears!

C. The wipers of the Alps pot are not directly connected to the grids of the ECC81, but via a Wima film capacitor. Why? Well, everything ages a little bit, especially tubes and volpots (and audio designers). So the theoretically worst case scenario is: The input tube (carefully selected) starts drawing a little bit more grid current than a new one (and the Alps pot has been used a lot). Then it could be possible that you hear some noises when rotating the volume knob, because the grid "current source" induces a small voltage on the pot's changing resistance. This is one of the few examples in which, in my opinion, an audio coupling capacitor is better than none.

D. I hope you had not too much trouble in measuring the high-frequency response. I use R+S UPD and B+K brands for general

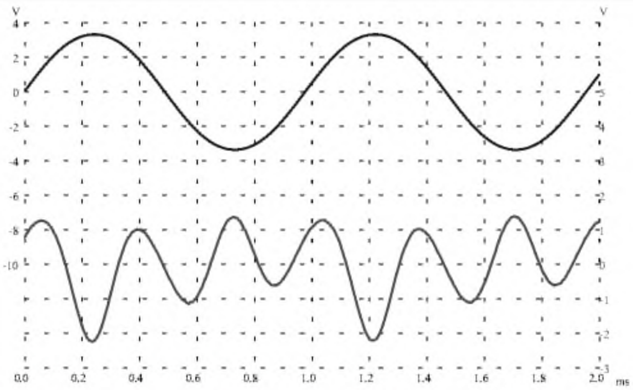

FIGURE 3: Residual distortion, 1k load.

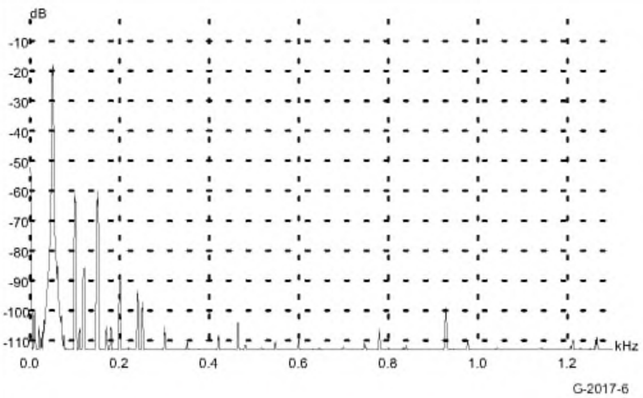

FIGURE 6: Spectrum of 50Hz sine wave, 100mV RMS, 35Ω.

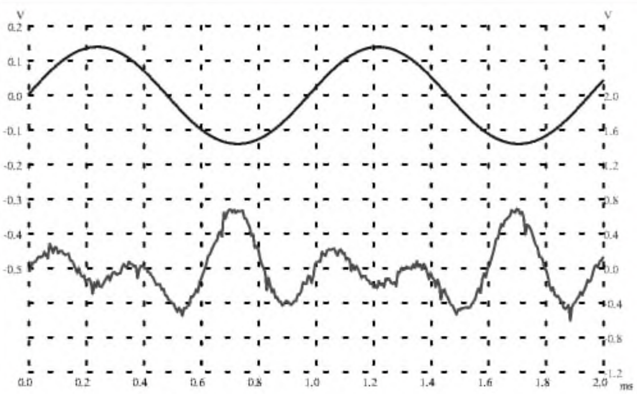

FIGURE 4: Residual distortion, 35Ω load.

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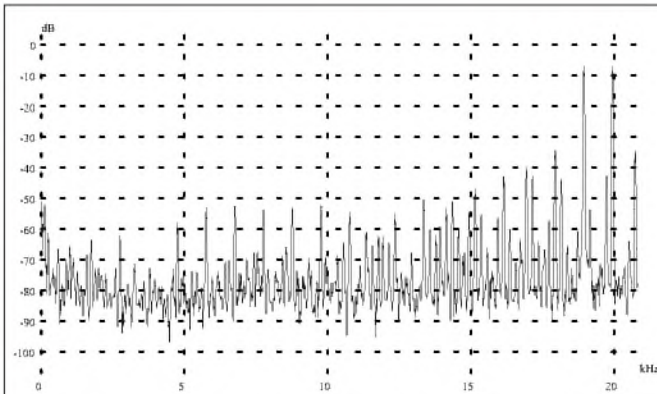

FIGURE 7: CCIF intermodulation distortion—0.283V p-p into 1k.

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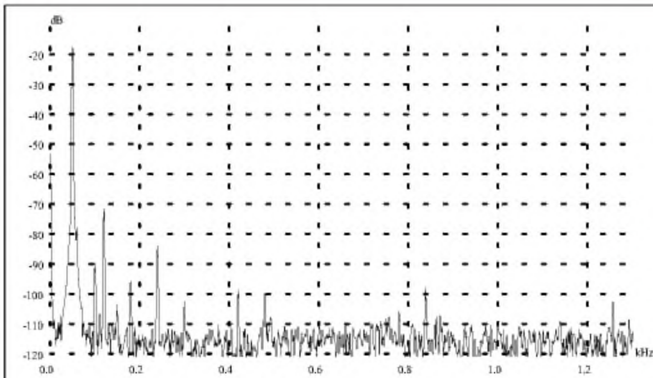

FIGURE 5: Spectrum of 50Hz sine wave, 100mV RMS, 1k.

G-2017-5

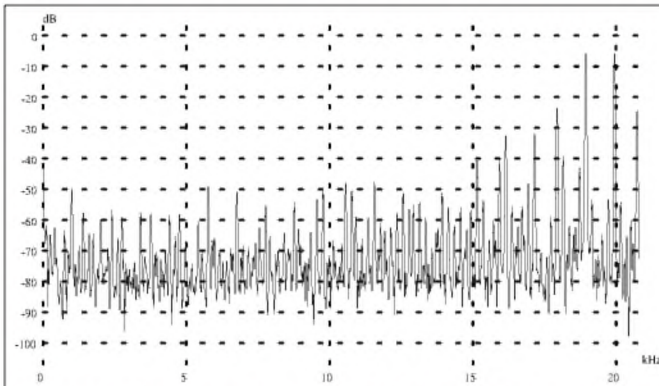

FIGURE 8: CCIF intermodulation distortion—0.283V p-p into 35Ω.

G-2017-8

audio measurements, and some HP devices for higher frequencies. Your result (530kHz on 35Ω load and 1.1MHz at 1kΩ) matches very well with the 1MHz/–3dB on 600Ω load that I stated.

The channel separation is a little bit less than what I measure, but I always shunt the non-operating channel with a 100Ω resistor to simulate a low impedance source (—60dB at 10kHz).

The resolution of your frequency-response measurement (Fig. 1) is so high that you can roughly calculate the output impedance of the EarMax from that. It is ~85Ω, as you have stated. This is quite different from my own, resulting in 37.5Ω. But it depends on the measuring method, the brand and condition of the ECC88s used, and the power-supply situation. You did, of course, use the U.S. power supply, made for 60Hz/125V nominal operation. I have opted for 125V as a safety margin because I am told that there are mains voltages up to 128V in some parts of the States.

I used an older set of 88s and a 50Hz/230V power supply driven with ~231V when measuring the output impedance. Additionally, I use a dynamic and not a static test method for that task.

My last point concerns the power output specification. It would be very easy to say: oh yes, it's a misprint; it must be 150mV instead of 150mW. Not so! It is definitely my mistake!

When I was asked years ago about that specification, I was very astonished because I found this the most unimportant spec of all, except for people who want to ruin their ears. Then I replied that it is definitely below 100mW for the EarMax and below 150mW for the Pro, having in mind that almost nobody knows how loud 1mW really is when listening with certain headphones. All I intended to mean was no risk of damaging your ears!

I had in mind that many of my friends in the business had damaged their ears in the 1980s, when transistorized headphone amps appeared on the scene that could easily supply ±15V or more even into low impedance (and efficient) headphones without clipping or any audible distortion. When I began designing EarMax and later the Pro, I wanted a device that could play loud—but not too loud—and give an easily detectable "soft clipping" distortion before anything could happen to your ears. Thank you for stating that just 100mV into 32Ω of your Grado headphones is uncomfortably loud (= 0.31mW). I hope that the extremely careful THD measurements you did prove that I was successful. If it is OK with you, I will give your power output/THD diagram as a reference in the future instead of xmW into yΩ.

The resolution of your distortion spectrum analysis is really great. I am unable to do the same (although my R+S UPD can easily do the –120dB) because our mains frequency is 50Hz. I actually had to use a 60Hz Sinus, and thus the picture I get is, in a way, the "reverse" of yours.

Thank you again for the extremely detailed and sophisticated job you did measuring the EarMax Pro. Not many technicians would pay so much attention to such a tiny and seemingly insignificant device.

Thank you very much for the great listening test you gave our EarMax Pro. I want to cut my answer to this part of the test very short. I can agree with nearly every detail of your description and have the same feelings when listening to EarMax—not as a technician but as an amateur musician and music lover. I play double bass just for fun.

I like most your comment that it's not perfect but I like it very much and I highly recommend it. That means to me that I'm, of course, not perfect as a human being or a technician, but maybe I have given the (thousands of) EarMax and Pro owners something to enjoy.

Stefan Brocksieper

Manufacturer's Response.

The gentlemen Hansen and Kastanovich must be highly commended for such a technically thorough and enthusiastic review of the EarMax Pro headphone amplifier. From my perspective I dare say that it is most rare to find an affordable product which endures the ups and downs of high end audio. Today's incessant question "What's new?" and the psychological need for the up-to-date, the latest greatest gear hardly apply to the EarMax and the Pro. Introduced at the 1994 Winter CES it has ravished the ears and hearts of a couple of thousand music lovers around the world.

Audio Advancements LLC is gratified to represent these wonderful products from Wuppertal, Germany, made to convey musical truth which we know remains unchanged and unchanging. Again a hearty thank you.

Hart and Beth Huschens
Audio Advancements LLC

Titanic 10″ Subwoofer Listening Critique

By Julie and Ken Ketler
Testing by Joseph D'Appolito

The story goes something like this (some names may have been changed): One particular afternoon while the Audiophilio family are cleaning out their basement, Dad finally summons the courage to ask Mom. Faun stares deeply into Francois' eyes as he approaches her, gently removing the box of holiday decorations she is holding. Trembling, nervous, and perspiring slightly, he takes her hand and whispers, "Sweetheart, listening to music and reviewing audio equipment with you has been nothing short of breathtaking."

Bending down on one knee he continues, "Will you put together a subwoofer kit for our next review for *audioXpress*?" As if she'd been longing to hear him ask this question, she quickly replies, "Yes!"

The whole basement becomes startlingly radiant, as a gentle breeze surrounds them both, blowing gently through her soft, light hair and across his broadening bald spot. The orchestral music swells into full fortissimo as they fall in love with each other all over again. They hold one another and sob gently. The dream of Francois Audiophilio, our friend, the amateur speaker builder, has now come true. He has found a woman who is truly willing to build sound equipment.

Note: Please pardon any hint of chauvinism here, but DIY (do-it-yourself) audio is a male-dominated hobby. First and foremost, we are here to thoroughly test this product and give you, the reader, as much insight as possible. Perhaps along the way, it will also be interesting to see whether a woman can find happiness in assembling a speaker.

IN THIS CORNER, WEIGHING 46 LB, INTRODUCING TITANIC SUBWOOFER!

Available from Parts Express, the 10″ Titanic Subwoofer System Kit is intended for anyone who wants to assemble his or her own subwoofer. Parts Express claims that even an inexperienced builder can put this sub together in about one hour! This seems to make the 10″ Titanic Subwoofer System Kit a perfect project for the speaker-building newcomer.

Yes indeed, but what about the quality of the finished product? Here's a hint: Vance Dickason (author of *The Loudspeaker Design Cookbook*) designed this system. Whoa! That alone speaks volumes about what we're dealing with here. Although this sub was designed with simplicity in mind, high performance was definitely Mr. Dickason's top consideration.

This kit (Parts Express #300-739) consists of a Dayton Titanic 10″ single-coil subwoofer (Parts Express #295-410), a subwoofer amplifier (Part Express #300-794), a compact (14¼″ W × 14¼″ H × 14¼″ D) ¾″ MDF acoustic suspension cabinet, which is solidly constructed, well braced, and finished with an unassuming pleasant black texture. Also included is a large sheet of eggcrate-style acoustic damping foam, spiked cabinet feet, and all necessary mounting accessories.

The Dayton 10″ driver is a very solid unit with a large magnet, heavy aluminum basket, polypropylene cone, and convenient spring-loaded connectors. The amplifier, which is rated at more than 250W into a 4Ω load, is designed specifically for subwoofer use and provides a variety of ways to integrate the 10″ Titanic subwoofer into your existing system. Here are some possible configurations:

1. If your current audio setup doesn't incorporate an active crossover, you can add the Titanic sub while running your main speakers full range (i.e., no high-pass filtering). For this setup, leave your main speakers connected as you'd normally have them. Run a stereo line-level output from your amp/preamp directly into the line-in connectors on the Titanic Subwoofer System amplifier.

   You can utilize many useful features of this sub, including a 12dB/octave low-pass filter (variable between 40 and 160Hz) along with a phase control (variable between 0 and 180°) to obtain a smooth transition between your stereo speakers and the Titanic sub. This configuration also provides a defeatable auto on/off power switch, channel summing, an infrasonic filter (for increased power handling), and an active 6dB boost at 30Hz (to fill out the bottom end of the audio spectrum).
2. Getting a smooth crossover response depends (as always) on your loudspeaker/room response, as much as electrical filtering. If running your main speakers full range doesn't allow smooth integration with the Titanic subwoofer, try the following setup: Connect the speaker outputs of your stereo amplifier to the high-level inputs on the Titanic sub. Then connect your main speakers to the high-level outputs on the Titanic.

   In addition to the features mentioned in configuration 1, this setup provides a 6dB/octave high-pass rolloff at 125Hz for 8Ω main speakers. Can you say "big capacitor"? Whew! This may make some high-end audiophiles cringe, but give it a listen. It just might sound great! What's 160µF between friends, anyway?

3. If you plan to use an outboard active crossover, you can use the internal amp of the Titanic sub for a bi-amplified system. However, the signal from your crossover will also pass through the Titanic's low-pass filter. Too bad it isn't defeatable! You could use them in conjunction with one another or consider the no-amp-included "passive" version of the Titanic Subwoofer System (Parts Express #300-740).

Before we could listen to any of these configurations, Julie had to assemble the system. Standing by with a full report, here's *audioXpress*' Julie Ketler. Take it, Julie.

T-NUT IS A FOUR-LETTER WORD!

So, he asked me to build a subwoofer. I knew it was only a matter of time. I mean, I am married to an audio enthusiast, a stereophile, and a real soundman. Is that big o' me?

At first, I agreed to co-write some reviews for *Glass Audio* magazine (my father was so proud—he, too, is one of you sound nuts). Now I'm putting together a subwoofer (I even know what one is)! Ken never would have asked me if he thought I couldn't do it, so I agreed to try.

I was somewhat apprehensive about it, but as soon as the 10″ Titanic Subwoofer System Kit arrived on our doorstep, I put on my ripped jeans and stained shirt and tied my hair in a ponytail. Then I went to work. I felt like a cool tool woman there in the cellar, surrounded by my own subwoofer kit—the first kit I ever put together. I admit, I was excited!

The assembly directions were fairly easy to follow, even for a novice like myself, but they were also somewhat foreign to me. I was not versed in the hardware terminology, so I had to ask my husband (who was only a few feet away, watching in delight) what T-nuts were. Also, stuffing the foam into the box was a full-body project; I had to push the gray foam in while standing above the box, using both hands to keep control of it.

■ TEST REPORT

By Joseph D'Appolito

For those of you who have followed my earlier loudspeaker test reports in *audioXpress*, this report will be quite a departure from what you have become accustomed to. Testing a subwoofer presents a very different problem from that of a typical full-range system. The low frequencies produced by a subwoofer challenge even the best anechoic chambers. Without an anechoic chamber, I will rely on ground-plane[1] measurements to characterize the subwoofer's acoustic performance.

FREQUENCY RESPONSE

The Titanic 10″ subwoofer uses a Dayton T1000 driver in a sealed enclosure with a built-in 250W amplifier and an active low-pass crossover. The crossover slope is 12dB/octave. The crossover frequency can be varied between 40 to 160Hz. The crossover cannot be bypassed. This is unfortunate because most home theater receivers and processors have a bass management system incorporating the 24dB/octave low-pass filter called for by the THX specification.

Frequency response was measured using the ground-plane technique. Drive level was constant throughout this test and set to produce a maximum SPL of 90dB at 1m with the crossover set to 160Hz.

Figure 1 shows subwoofer response at crossover settings of 40 and 160Hz. A third curve shows the response with the variable crossover frequency control set at its midpoint. Although no intermediate points between 40 and 160Hz are marked on the control, the midpoint setting appears to correspond to a crossover frequency of 80Hz. Notice that the peak response level varies with crossover frequency. When adjusting the crossover frequency, you will also need to change the level control to maintain proper balance between the subwoofer and the main loudspeakers.

Table 1 lists the −3dB and −6dB frequency for each setting of the crossover frequency. These points are measured relative to the maximum response level for each curve. With a 160Hz crossover, the −3dB bandwidth extends from 53 to 110Hz. The −6dB point is 41Hz.

In the octave below 41Hz response falls 15dB. This is the half-space response. In smaller rooms, room gain may offset much of this drop in response.

With an 80Hz crossover, the −3dB bandwidth extends from 46 to 96Hz and the −6dB point is 37Hz. In the octave below 37Hz response falls 16dB. Corresponding figures for the 40Hz setting are 39 to 90Hz, 32Hz and 18dB.

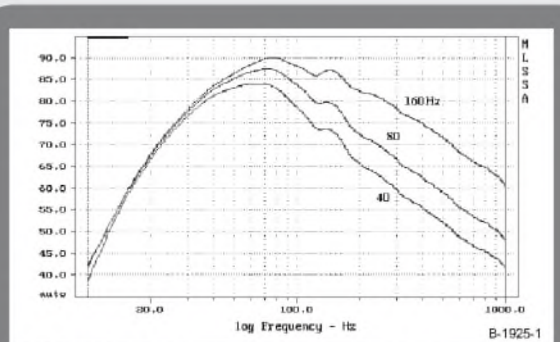

FIGURE 1: Titanic 10 frequency response versus crossover frequency setting.

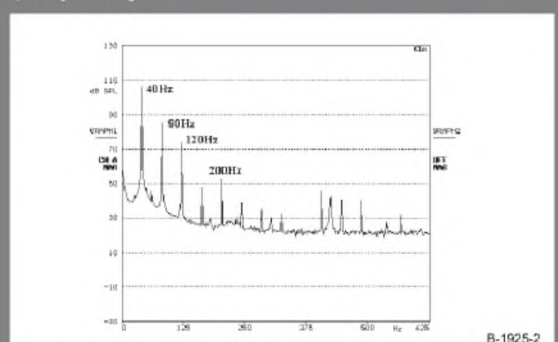

FIGURE 2: Harmonic spectrum at 40Hz and 94dB.

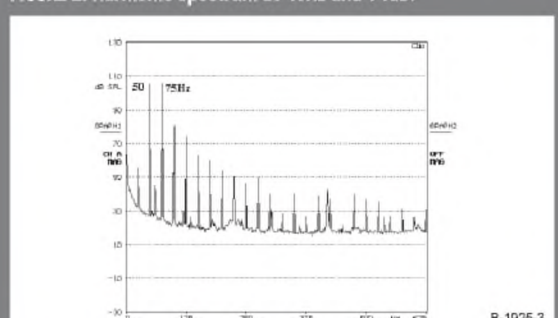

FIGURE 3: IMD spectrum at 96dB, with 50 and 75Hz frequencies.

MAXIMUM SPL AND HARMONIC DISTORTION TESTS

Maximum subwoofer SPL capability is of great interest. One common way of rating the subwoofer is to measure the SPL at 10% harmonic distortion.

A test was conducted at frequencies between 20 and 60Hz in 10Hz increments. At each frequency, output level was increased until 10% total harmonic was reached. The results are shown in *Table 2*. At and above 60Hz the THX home theater specification of 105dB was reached with less than 10% distortion.

Ken was laughing when he said, "Just stuff it, Julie," reassuring me that he wasn't being rude. The instructions didn't mention how much foam to use, so I used all that was supplied.

The other steps were fairly smooth, and I was able to follow them without asking any questions from my admiring mentor, until it was time to set up the input and output connectors. The directions for these were slightly advanced for me, and I would not have been able to put them in correctly (which I'm sure could be a disaster) if I didn't have some help.

The last step was to attach the woofer to the enclosure. This would have gone flawlessly, but some T-nuts kept popping out and magnetically attaching themselves to the woofer. This was a little frustrating, because I had to keep removing the driver and readjusting the T-nuts over and over! Again, I am a novice and an admitted mechanically challenged individual.

All in all, I think I did pretty well, considering my status! We looked over the final product and I was quite proud of myself. As we set up the Titanic sub in our listening area, I felt a real connection to it (no pun intended, really!)

If my dear husband asked me to put together another kit, I would definitely say "yes." Of course, the moment would have to be perfect and the lighting "just so!" Wives of audiophiles can either fight husbands or join them. I choose to join mine, and I'm very glad I do.

RESULTS

The highest hurdle that we had to leap was matching the sub to our main speakers. Although the frequency and phase are adjustable on the Titanic Subwoofer System, the slope of the filter is fixed at second-order. Theoretically, this seems far more limiting than it actually is. You might be inclined to believe that a fourth-order vented design couldn't possibly mate very well with this sub. We tried this with a few pairs of speakers and realized that it was quite possible to match the timbre of the sub with vented main speakers, given the effects of room acoustics.

In our toughest case, we used a pair of vented enclosures with 4″ drivers, which drop off sharply below cutoff. We tried configurations 1 and 2 to find the best compromise. Using configuration 1, the crossover wasn't tonally seamless, but okay nonetheless. In configuration 2, the overall tone of the system was more consistent, although bass definition diminished somewhat. Due to the high frequency/low order filter, the location of the sub was audibly evident.

With our best case, we used a pair of homemade enclosures, each housing a 5¼″ woofer and providing healthy response down to about 50–55Hz. We easily tuned the Titanic filter, which gave us a very smooth transition from mains to sub. The subwoofer added wonderful body to the music that just seemed "right" without calling attention to itself.

On one occasion, Ken's Dad mentioned that he couldn't hear the sub and didn't think the power was turned on. When the bass and drums of the tune began to play, he shook his head, "Oh yes, there it is!"

Our unsolicited opinion is that new subwoofer owners may often be inclined to turn the sub level up too high in an effort to "hear" their new baby shake its stuff. Initially, we did this, too, but we quickly became over-bassed and lowered the level of the Titanic sub.

A quick check with the *Hi-Fi News & Record Review Test Disc #3* and our Radio Shack SPL meter helped us fine-tune the bass level. Once we were finished with test tones, we were ready to listen to some music. We decided that configuration 1 seemed as though it might be the most common way to hook up the sub, so we conducted our "official" listening with this setup.

TEST TRACKS

John Patitucci—"Showtime" (from Another World GRP grd-9725)

KK: This contemporary jazz track is made up of three simultaneous

At 40Hz 10% harmonic distortion corresponded to 94dB SPL at 1m. This is a half-space value. Corner placement may increase this number by up to 9dB. The subwoofer output spectrum at this level is shown in *Fig. 2*.

Significant distortion components can be seen at 80, 120, and 200Hz. The 106dB level on the plot results from the ground-plane measurement at 0.5m. This corresponds to 94dB half-space at 1m.

For home theater application, SPLs of 100dB below 80Hz are commonplace. The Titanic's harmonic distortion at these SPLs may seem high, but the audible impact at these low frequencies is probably not as significant as the numbers alone would imply.

A home theater system meeting THX specifications must be able to produce SPLs of 105dB. Even with corner placement and room gain, the Titanic 10 will not meet this specification below 40Hz.

TABLE 1
TITANIC 10 FREQUENCY RESPONSE VERSUS CROSSOVER SETTING

CROSSOVER FREQUENCY (Hz)	PEAK RESPONSE (Hz)	–3dB (Hz)	–6dB (Hz)
160	77	53	41
80	72	46	37
40	65	39	32

TABLE 2
MAXIMUM SPL FOR 10% THD VERSUS FREQUENCY

FREQUENCY (Hz)	MAX. SPL/1M (dB)
20	78
30	80
40	94
50	98
60	105 (4.8%)

INTERMODULATION DISTORTION

A two-tone distortion test was run using frequencies of 50 and 75Hz at a 96dB level. As shown in *Fig. 3*, this test produced a very dense number of harmonic and intermodulation distortion components. However, most of these components are harmonic distortion. Total IMD is only 2%.

A Note on Testing: The Titanic 10″ subwoofer was tested in the laboratories of Audio and Acoustics, Ltd. Measurements were made with the MLSSA and CLIO PC-based acoustic data acquisition and analysis systems using an ACO 7012 ½″ laboratory-grade condenser microphone.

MANUFACTURER SPECIFICATIONS

Designed by Vance Dickason and featured in 6th Edition *LDC*—14¼″ W × 14¼″ H × 14¼″ D, 46 lb—¾″ MDF cabinet, 10″ Titanic driver, textured black polymer coating, coupled to floor with "black chrome" spike set. When coupled with Parts Express 250 W amplifier: 112dB maximum output, f_3 of around 24Hz (room loaded), 30Hz bass boost, and frequency response 24–160Hz.

REFERENCE

1. J. D'Appolito, *Testing Loudspeakers*, Chapter 4, Audio Amateur Corporation, Peterborough, NH, 1998.

CONTACT INFORMATION

Parts Express
725 Pleasant Valley Dr.
Springboro, OH 45066-1158
513-743-3000
1-800-338-0531
FAX 513-743-1677
e-mail: sales@partsexpress.com
website: www.partsexpress.com

bass parts (yes, three) including 6-string, 4-string, and piccolo bass. On top of those is a great deal of drums and other percussion. The Titanic sub does a very nice job of keeping each bass (including the bass drum) very distinct and sharp with a huge kick in the gut when we crank it up to 11!

JK: Will Kennedy's drum set sounds magnificent with the Titanic. Without it, the drums sound a bit distant and therefore not as enjoyable. The Titanic subwoofer seems to complete the musical selection—truly a great addition to any speaker system.

"Beethoven Symphony no.3" in E Flat, op.55 "Eroica," London Symphony Orchestra, Wyn Morris: Conductor (MCA mcad-25237)

KK: All four movements of this symphony are nice demonstrations of the Titanic sub. There are some quiet sections that have barely audible bass violin plucks and long droning notes in which the Titanic fills out the bottom end of our main speakers, lending a calm, yet powerful foundation.

JK: With the Titanic subwoofer, the tympani drums sound very low and punchy. This adds real emotion to the crescendo sections. As they play low, I get a real sense that I am in a big concert hall rather than our living room.

Pink Floyd—various tracks (from *Dark Side of the Moon* MFSL UDCD-517)

KK: The disc starts with a track called "Speak To Me," which is a heartbeat-like effect fading up into a wash of mind-boggling extraneous noise. I've heard this countless times before on many people's systems. With the Titanic sub, I sense this heartbeat before I'm aware it's coming from the CD. Other systems I've heard present this "thump-thump" along with a nondescript rumble from their respective subs that sound bassy but unfocused, losing this ominous effect.

"Us and Them" starts out as a fairly gentle song with light drums. The bass drum, however, plays an integral part in producing the meandering mood of this number. The Titanic sub does a fantastic job of transferring this subtle impact to the listener.

JK: When we add the subwoofer to the main speakers, I can hear a subtle difference in the all-around sound. I think it's more of the way it makes the music "feel" rather than "sound." The Titanic subwoofer makes this entire Pink Floyd disc richer and altogether better than running just our main speakers.

SUMMARY

Overall, the 10″ Titanic Subwoofer System Kit is absolutely super, especially when you consider it costs a mere $350 from Parts Express! We found that the 10″ Titanic Subwoofer System has very smooth and "quick" response in our listening room. It performs very cleanly even at relatively high SPLs. The variable active low-pass filter, phase control, and passive 125Hz high-pass filter make this system quite versatile and easy to integrate with little to no amount of grunting and fussing from the user.

The 10″ Titanic Subwoofer System Kit is quite assembly-friendly. However, since for many a kit is often the first step into the speaker-building world, the Titanic sub could truly be a perfect learning tool if the assembly manual briefly described all of the parts and why each is necessary—perhaps a precursor of a "how to" book. It may also be helpful to add examples of different system configurations and how to integrate the sub.

When all is said and done, apparently a woman can build a subwoofer and even enjoy it, leaving behind many June Cleaver-isms. As the Audiophilio family builds more DIY equipment in their basement, perhaps Francois can run upstairs for lemonade, while Faun inhales solder fumes and shouts, "Here, honey, hold this" on a more regular basis. Francois seems thrilled with this idea, but for undisclosed reasons he seems a bit hesitant to wear high heels and a housedress. ❖

AC POWER LINE

Regarding the article "The AC Power Line and Audio Equipment, Part 3" (February '02), page 35, "Power Conditioners and Filters," in short, "I don't think so."

The author's analysis is based on a very short connection between the service panel and the EMI filter. In real life, there are many feet of wire between the service and the filter. At that point the white "neutral" wire at the input to the filter is almost at the same impedance above ground as the black "hot" wire. The longer the wire, the more the black and white wires have common mode rather than differential mode RF on them.

The inductors in the filter present some reactance to RF. This greatly reduces the amount of RF current that flows through the filter, so the path shown in Figure 17 from the center tap of the second set of capacitors will have very little, if any, RF current to ground. Also, since the RF level on both the black and white wires will be very similar in practice, there will be little differential RF voltage between them and thus little RF through the input capacitor.

In real-world applications, little RF current flows in differential mode on AC power lines. The exact amount depends on the length and the wiring style. Old-style "knob and tube" may have very different "hot" and "neutral" wire paths. Modern conduit, or "Romex," wiring has the two wires very close to each other.

In general, my experience is that any noise detected on "ground" is from ground faults—either device failures or improper wiring. The most common example of improper wiring is a neutral-to-ground connection at locations other than the single required neutral-to-ground bond at the service panel.

Whenever this happens the "ground" conductor shares current with the neutral and, due to real-life resistance in wire, ground potential differences exist. If the load is severely nonlinear (such as a switching power supply or a variable-speed motor drive), there will be harmonics of 60Hz on the line between "hot" and "neutral," but, if there are no ground faults, little noise current in "ground."

Three-phase systems have a well-documented problem with excessive neutral current due to nonlinear loads. Modern three-phase distribution systems oversize the neutral by at least one wire size to handle those nonlinear harmonic currents. Again, no current flows through ground. Few homes have three-phase power distribution, though.

One very easy test for ground faults is to measure the current in the wire between the service panel ground connection and the earth. There should be absolutely zero current flowing through this wire. (I've seen 20 and 30 amps at times!) In a well-installed and operating system there is no potential difference between any two ground connections. (I have seen more than 10V AC, too.)

Second, in my experience I have seen no improvement when "balanced power" is used. My experiments have shown that an isolation transformer may reduce (and rarely eliminate) ground fault currents, but I have found

no difference between grounding the center tap or the "neutral" connection on the secondary of an isolation transformer. Very fancy power conditioners, in my opinion, do more to reduce your cash flow than to improve the quality of AC power provided on their output.

Finally, on page 36, Figs. 18–20, the ground connection on the balanced power secondary is not connected to the ground connection on the service panel. This is a violation of the National Electric Code. There can be only one ground reference in any power system, and that ground reference must be the one that bonds the "neutral" to "earth" at the service entry panel. Violations of the NEC can be serious safety hazards as well as giving an insurance investigator grounds to void policies.

It is permissible to have a much greater than minimum code ground system. I routinely specify those at communications sites. The inspectors shake their heads at the "wasted" money to have a low impedance ground system, but they must accept it.

Remember, there are parts of the world that do not follow the "North American" power distribution plan. Many of these areas operate with ungrounded neutrals. This makes a big difference in design and power-supply noise analysis.

Bill Ruck
Broadcast Engineer
San Francisco, Calif.

Charles Hansen responds:

Mr. Ruck raises some good points in his letter. My article was not so much an analysis as an overview. The short connections Mr. Ruck refers to are, as he mentions, dependent on length and wiring style. In my home, the electrical wiring from the breakers is routed together either through the attic or basement until each feed in turn breaks out of the bundle to proceed towards its destination circuit.

Also, "length" is very frequency dependent. Conducted EMI measurements begin at 150kHz, where the wavelength is 2,000 meters. I have been involved in troubleshooting grounding and EMI problems in our electrical laboratory, and on aircraft from Lear Jets to Boeing 747-400s.

Electrical wiring does not have a controlled impedance like coax cable. There are times when I have seen vastly different levels of EMI on the power feed and its neutral return, using a clamp-on RF current probe and spectrum analyzer. I have even seen RF currents on the steel rudder and elevator control cables. The interfering source was often the aircraft HF transmitter (3–23MHz). There is very little "always" and "never" with EMI, more like "usual" and "uncommon." EMI does whatever it wants to do.

I apologize for my shorthand use of the ground symbol in Figures 18–20 instead of showing the actual ground conductor. It was my intent to show the secondary ground always being returned to the electrical panel. Mr. Ruck is absolutely correct about the NEC grounding requirements.

I read Mr. Hansen's informative and interesting article on AC power quality (Feb. '02), which offers a great wealth of information. However, there is a safety issue concerning MOVs used in surge suppressors that I am compelled to bring to your readers' attention.

I work for a large manufacturer of surge suppressors and previously worked for Underwriters Laboratories Inc. (UL) for several years. Mr. Hansen implies that an overcurrent fuse alone is adequate to protect MOVs in a surge suppressor. This is not true. Overtemperature protection is also required.

There are several situations in which a sustained overvoltage/limited current condition can occur on the powerline. This causes the MOVs to conduct current continuously and overheat. A 20mm MOV at 5A continuous current will turn "cherry red" and can ignite the suppressor enclosure or nearby combustibles. An 8A overcurrent fuse will not prevent this event. You should also note that currents of less than 1A can cause an overtemperature failure in some surge suppressors. In 1998 UL revised their safety standards to require overtemperature protection. Readers who may want to build their own surge suppressor should make a note of this safety issue.

James Forte
Framingdale, N.Y.

Charles Hansen responds:

Thanks very much for this important safety update. There is some information on the *web about UL 1449, although UL charges $30 for copies of their safety standard.*

In the third part of Charles Hansen's article, "The AC Power Line and Audio Equipment" (Feb. '02), the definition of an "on-line" UPS unit as printed is incorrect.

The inverter in an on-line—or "true"—UPS unit always obtains its power from the battery which is trickle-charged by the connected AC. In the event of a power disturbance, the charging may be interrupted but not the downstream supply of inverted power. One of the advantages of this type of power supply is that it inherently isolates the equipment to be powered from the vagaries of the public utility.

These units don't switch over so they shouldn't suffer from the harmonic problems that Mr. Hansen says are a result of that circuitry. I suspect that the AC generated by these units would often be much better than that provided by the utilities. Mr. Hansen, perhaps you could elaborate on why you believe this type of unit would not provide "clean" power.

On a different but related topic: If an audiophile is willing to spend the money on a fancy shielded power cord, you might assume that he would also have considered providing a dedicated circuit or circuits to his equipment. This candidate audiophile should have his electrician run wiring in rigid conduit. This method would likely provide the level of shielding required to make the expense of the power cord worthwhile. Has anyone had this type of wiring installed in their listening room and then tried any of these exotic power cords?

On a different but unrelated topic: Many years ago I read with great interest A.R. Bailey's articles on the non-resonant loudspeaker cabinet—the transmission line. When he described his goal of "losing the backwave" the association I immediately made was with the infinite baffle concept. Over the years I was surprised that no one else seemed to make this connection. While scanning the article "Infinite Box Concept—Part 2" (Feb. '02, p. 38), I experienced a little déjà vu.

Although the authors discuss "transmission line-like" effects, such as the attenuation and phase shifting of the backwave, they only compare their design to vented and sealed box designs. It seems to me that some useful insight might be gained by comparing these two concepts. Just a thought.

Mark Henschel
Quadrangle Architects Limited
mhenschel@quadrangle.ca

Charles Hansen responds:

I researched Mr. Henschel's information concerning "on-line" UPS and, alas, I found it defined as he described, as well as with the definition I used in the article. Thanks for the information.

The fact that a UPS does not switch over to provide emergency power is not the issue with harmonic generation. Maximum efficiency is the primary mission of a battery-powered UPS, and a low-distortion sine wave is not easily generated with high efficiency and low cost.

Mr. Henschel's suggestion to run a dedicated AC feed in steel conduit is a good one. If a steel outlet box is also used, this will provide both E-field and H-field shielding right up to the point where the power cords are plugged into the outlets. Depending on the accessibility, conduit may or may not be an easy retrofit in an existing listening room.

GLORY DAYS

Just wanted to tell you how much I am enjoying my copy of the *Audiocraft* reprint volume and am looking forward to the sequels if you elect to do them. It brought back a lot of memories and prompted me to look up one of the authors, Glen Southworth. I met Glen in the 1980s when we were both in the teleconferencing industry. He evidently wrote a lot of articles for *High Fidelity*, *Audio Engineering*, and *Audiocraft*. He is still involved with leading-edge video technology and now lives in Colorado.

I think *audioXpress* is well balanced and finely written. It underscores that your organization is the last bastion of the technical audiophile. I hope you can keep it going for years to come.

Ken Bird
Autogenics/Stoelting Polygraph
Wood Dale, Ill.

[*The* Audiocraft *reprint is available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, 888-924-9465, www.audoXpress.com—Eds.*]

LOAD VALUES

I had the opportunity to view a recent magazine (Jan. '02) and was interested in the article entitled "A Beginner's Push-Pull or Single-Ended Amplifier." I noticed that this design was similar to an amplifier that I designed a few years ago, but with some areas corrected in terms of driving the output transformer. As you are most probably aware, the 6SN7 valve is of the small signal variety, where the minimum recommended load for a single element is 47KR, and by paralleling both halves of a single 6SN7 valve, the loading should be 22KR, and not 3.5KR as shown.

Although it is nice to think this valve would work, to be fair, the reverse would be true with a distortion that would be totally unacceptable to anyone who designs or builds valve amplification. I would therefore recommend that adjustments be made to the A-A increasing this from 3.5KR to 22K, which would at least put the valve in its correct operating conditions and provide something more realistic as to the excellent sound that these valves in this form of application can provide.

Chris Found
CFDesigns (UK)

Larry Lisle responds:

Thank you for writing. The usual practice when using a 6SN7 as a small signal voltage amplifier is to use a plate resistance of 47kΩ or more. In this circuit the 6SN7 is being used to deliver power to a load, and it's customary to make the load twice the resistance.

While increasing the load impedance will indeed lower the distortion, the advantage of a slight decrease in distortion is outweighed by a rapid fall-off in output power. May I refer you to the Radiotron Designer's Handbook, *4th Edition, Chapter 13, Section 2, part (iv) for a graph of these factors for a type '45 tube.*

The "twice the plate resistance" rule is only a guide to a good compromise, and I've built good-sounding amps with the plate load both above and below this figure. The important thing is to experiment, challenge the conventional, and see what sounds good to you—then write it up for audioXpress*!*

SOURCE

As for those otherwise impossible-to-get Toshiba FETs Mr. Borbely likes to use, MCM Electronics lists the BA1404–used in the Ramsey FM10.

John Nickerson
Chatham, Mass.

MORE I^2S OUTPUTS

Thank you for printing my letter "I^2S Outputs" in the Nov. '01 issue. As it turns out, Burr-Brown makes a demo board (DEM-DAI 1704) that is somewhat similar to the "I^2S Mate" I proposed (*Figs. 1* and *2*). I think it costs about $350.

I haven't bought this board, but it appears you can directly put an I^2S signal (16 or 24 bits) into the excellent DF 1704 filter through the CND2 connector. I don't know what this connector looks like and would prefer to use a rugged, bipolar device (LS) as a receiver, but with care and, of course, proper selection of the logic options presented by the DIP switches, it appears a resourceful amateur could get an I^{2}S signal directly into DF 1704.

The DEM-DAI 1704 does not have ground-plane isolation between the filter and DAC chips I proposed, but honestly I don't know how much or even whether that results in better sound: Given my druthers, yes, I would incorporate ground-plane isolation. It's my present feeling the subjective sound of 16/44.1k digital audio can be improved by "babying" the interpolation circuitry.

The Pacific Microsonics PMD-100 filter in my Camelot Arthur improved with OS-con bypass caps, a heavy layer of Music Coat (nasty fumes) and a Shakti-stone bonded on top. Individually these were quite subtle improvements, but taken together I think it's been a worthwhile endeavor. I think I could get another improvement with a passive non-feedback regulator.

I believe that the big advantage of Sony's SACD product is the lack of interpolation circuitry. If you could get the interpolation circuitry of CD players to work better, most of SACD's subjective advantage would disappear. I'm not jumping on the SACD bandwagon. Some of my CDs sound incredible–closer to the master tape sound than I ever got with LPs.

Rick Bergman
Missoula, Mont.

P.S. This device doesn't have a 20-bit I^{2}S setting. I actually wrote to T.I. and proposed making DF 1704 in a larger 44-pin (greater pin spacing) surface mount package (as is PMD-100) along with a 20-bit I^{2}S in an interpolation only output (no filter, but that might be impossible), and balanced outputs for those that need them. Fat chance they'll make it, but at least I tried. My DTI Pro-32 sounds best at a 20-bit setting. 24 bits sound "distant." DEM-DAI-1704 might work great with the Perpetual Technologies PA-1 (better word-length algorithm).

TV SWEEP TUBES

As an avid fan of sweep tube amplifiers (see my article, "The Snubber," *GA* 3/95, p. 18), I read Mark Tavernier's article in the Jan. '02 issue with interest ("A PL504 PP Amplifier," p. 42). Unfor-

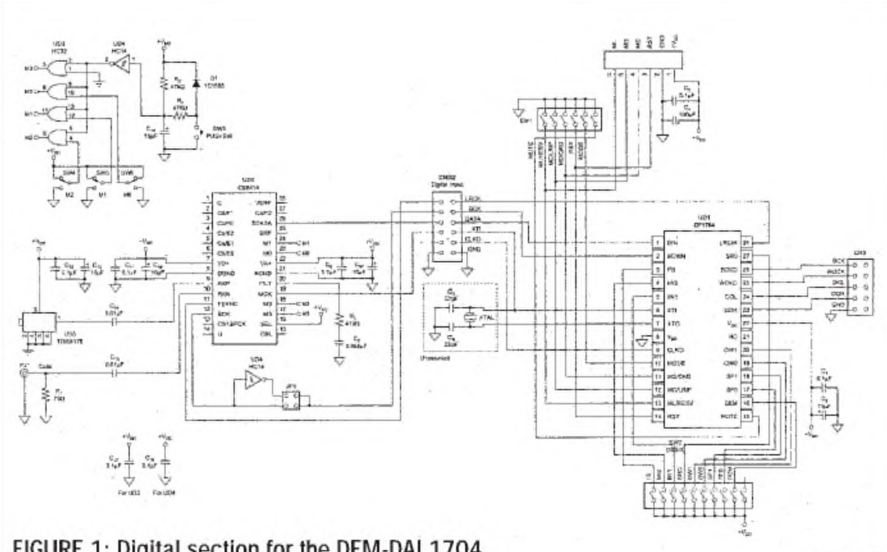

FIGURE 1: Digital section for the DEM-DAI 1704.

A-bergm-1

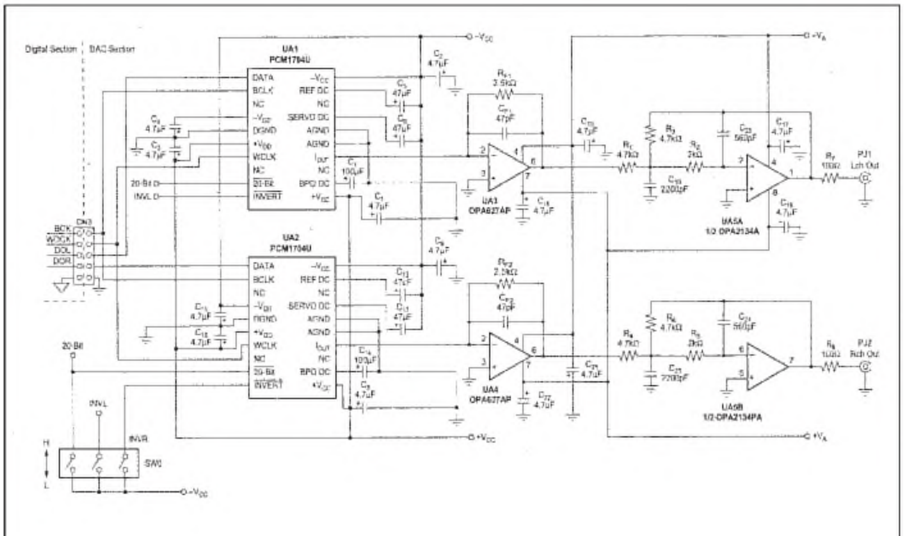

FIGURE 2: Analog section for the DEM-DAI 1704.

A-bergm-2

tunately, Mark is using the sweep tubes incorrectly in his amplifier.

Unlike audio beam power tubes such as the 6L6, sweep tubes utilize a very small spacing between the cathode and G1 and G2. Thus, the screen grid has a low peak voltage rating; one that is vastly exceeded in Mark's design. The result is inefficiency, distortion, and/or a short tube life.

In fact, when properly used, one of the PL509s will provide at least 15–20W of undistorted power in single-ended triode mode. This is done by connecting G1 to the cathode and driving the input signal to G2 via a direct-coupled cathode follower. A pair of the PL509s triode connected in push-pull will easily put out ten times the power of Mark's design, or 100W, provided that the output transformer and the plate supply are properly designed to handle the increased power.

Sweep tubes have a large, efficient cathode and close grid spacing, so they are able to drive low impedance loads almost to the lower rail of the supply. Sweep tubes have a large amount of current drive capability, so they have a lot of headroom. This shores up the low-frequency response as the output transformer efficiency rolls off. The amplifier thus has bass response similar to that of a solid-state amp, but the transcendent highs and midrange let you know that you are running tubes. That is why I like sweep tubes so much. But please use them correctly!

David Wolze
San Jose, Calif.

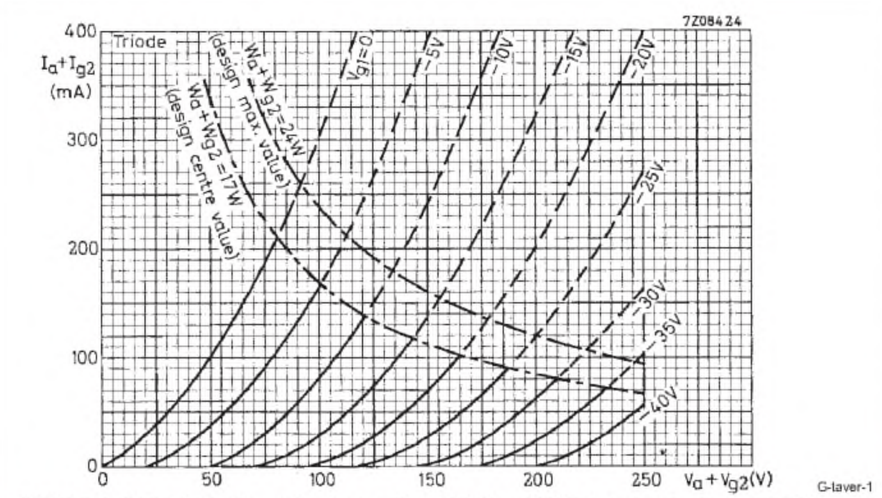

FIGURE 1: PL504 triode mode curves (from Philips MBLE data book, Sept. 1970).

Mark Taverniers responds:

Mr. Wolze's letter raises some interesting points concerning the use of TV sweep tubes (beam pentodes) in audio amplifiers. The PL509 he discusses does, however, present some differences compared to the PL504s used in my design.

	PL504	*PL509*	*[unit]*
Peak anode voltage	*7000*	*7000*	*V*
Maximum cathode current	*0.250*	*0.500*	*A*
Maximum anode dissipation	*16*	*30*	*W*
G2 voltage	*200*	*175*	*V*
G1 voltage	*-120*	*-10*	*V*

The PL509, which I have used in the past as a series pass element in HV regulators, is obviously a more powerful tube and would no doubt offer the kind of output power Mr. Wolze suggests. Part of my work was based on the Philips/MBLE curves for the PL504 in triode mode (G1 driven/G2 to anode), which unfortunately I do not possess for the PL509.

I was, however, unaware of the internal structure of this tube as Mr. Wolze describes. At the time of design I considered his suggestion to tie G1 to the cathode and drive the tube from a low impedance source using G2 (SPICE simulations based on the tube model described in the article would indicate some 65W in triode PP at less than 2% THD).

However, I would argue with Mr. Wolze on the following:

- Because, strictly speaking, the PL504 is a beam pentode, would driving it via G2 while keeping G1 at cathode potential make it a true triode?
- At the outset of the article, I stated that my goals were about 10W output power and the use of available components where possible. Going for the kind of specs Mr. Wolze suggests would have involved ordering custom-wound output and power-supply transformers and have brought the total project cost closer to $1000 as opposed to the $300 mentioned in the article.
- As you will ascertain from the enclosed graph (Fig. 1), THD is about 1.5% at Pmax, which I consider quite satisfactory for this kind of amplifier (Ui = 1kHz sine/8Ω resistive load).
- Since my first tests, the amp has been running for at least 1500 hours (I adjusted the bias approximately every 500 hours) and shows no signs of unreliability or premature aging. An Ip/Up curve plotted on one of the PL504s some weeks ago compared almost exactly with the ones I measured upon receipt of the new tubes about three years ago.
- Mr. Wolze has a valid point where it comes to inefficiency. I would need to invoke a kind of artistic freedom to defend a design that is barely 15% efficient, but this kind of freedom is certainly one of the aspects that makes our hobby worthwhile.

Nevertheless, I must thank Mr. Wolze for his critique and added insight into the use of TV sweep tubes. Given the right opportunity, I *would consider it a challenge to build a really powerful amp using these tubes and with his suggestions in mind.*

TRANSFORMER SOURCE

I enjoyed the well-written article on moving-coil phono step-up transformers by Pete Millett (May '02 *aX*, p. 58). Transformers are an overlooked source of gain in an audio system.

I have one bit of information to add. In addition to the transformers mentioned in the article, Lundahl Transformers AB (www.lundahl.se) makes two moving-coil step-up transformers: LL9206 and LL1678. You can connect the former for 14dB, 20dB, or 26dB of gain; the latter for 18dB, 24dB, or 30dB of gain. I have used the LL9206 in a push-pull phono preamplifier I constructed with help from Piedmont Audio Society members and find it to be an excellent performer. Two factors led me to choose this transformer: the connection versatility and the use of an amorphous core, which lowers distortion.

I purchased these from the Lundahl distributor in the U.S. Their website is www.kandkaudio.com.

Fernando Rodriguez
Apex, N.C.

REVIEWS WANTED

I have been a subscriber to your magazine in its current form and in previous forms for some time. I looked at your review of the Boston Acoustic Home Theater System (February '02), and I have to ask why you continue to review products like this that are reviewed on a regular basis in other magazines. Our hobby is such a fragile one, and so many good, committed people go out of business all the time–witness the recent demise of Tango transformers.

Why doesn't your magazine support people like this–people who will never be reviewed in other magazines, and who could benefit tremendously by a good review in this venue? Why not use the space to review a home theater kit from Zalytron, which I have been tremendously interested in learning something about for some time . . . and by the way, someone who also advertises regularly in your magazine? Or something from Parts Express, who is also in the kit business, or many of the other people who I see regularly advertise in your magazine.

It seems to me that if this hobby is going to stand any chance of surviving this time, the people who are involved must take a more active role in protecting each other. I am absolutely convinced that Boston Acoustics is going to survive just fine, thank you, whether or not you spend any time reviewing their products, but what about all of these other people, who more directly also influence your very survival. People who also offer a product like these kits, which are more in touch with the general editorial concept of your magazine, and which I bet stack up as good or better than these mass-produced products.

Gary Johnston
Princeton Jct., N.J.

Good point, Gary. You seem to have grasped the benefit of a favorable review in our publication better than our manufacturers. It may surprise you to learn that our repeated calls for manufacturers to send us products for review have been met with lukewarm acceptance. We will continue to solicit products—especially kits—in hopes that the right people get the message.—Eds.

HELP WANTED

I have recently acquired a quantity of sound barrier material: fiberglass-reinforced, barium-filled vinyl, which weighs approximately 1 lb/ft^2, and is similar in its barrier qualities to sheet lead. Would this material have a value as a liner for speaker cabinets? It would seem to be ideal in eliminating the resonance of the cabinet material, and for folded horns could provide a smooth surface for the inner curve instead of the "coopered joints" that many designs feature. ❖

Paul E. Clinco
paul-clinco@mindspring.com
5655 East River Road
#101-163
Tucson, AZ 85750

Readers with information on this topic are encouraged to respond directly to the letter writer at the address provided. –Eds

Weaver

from page 26

plates to the screen grids. You could probably connect them directly, but I think the additional isolation the resistors provide is a good idea.

Build the power supply first and test it under load. I always use a couple of junk-box power resistors connected in series for a value of about 5k across the output. This draws enough current to let you know it's working.

Build the rest of the amp and check it carefully for any wiring errors and connections you forgot to solder. Connect speakers or a proper resistive load to each output, turn it on, and measure the voltage across the cathode resistors of the parallel output tubes. It should be about 20V. Plate voltage should be 300V or so.

Expect some hum. There always seems to be some in any SE amp, no matter what you do. But it's low enough that I can't hear it from the listening position, and a little hum never bothered me, anyway. Don't bother with DC filaments, which make the amp a little quieter but don't sound as good as AC filaments, for some reason.

The result should be a very smooth and transparent-sounding amplifier, which, despite the low power, is not weak or thin-sounding at all. I give much of the credit to the choke-input design of the power supply.

The old 6L6G never sounded so good. ❖

MacArthur

from page 49

special meaning for you long before an emergency hits. Perhaps you would choose to take audio equipment and recordings, perhaps not.

Certainly Duncan and I never pondered this question until chaos erupted in our lives in the early hours of May 11, 2000. We found ourselves making rapid choices about which possessions to save while the radio urged us to leave town immediately. Thinking about these decisions earlier would have caused less stress. And we might have made better choices. ❖

Classifieds

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AD INDEX

Yard Sale

FOR SALE

JBL 108H mid-bass (2), $35 each; Audax HM170ZO 6.5" woofers (2), $30 each; Audax HM100X2 midrange (2), $20 each; Audax AWO25S3 tweeters (3), $17 each. Ernie Holmes 916-726-7039, e-mail ernieh@lanset.com.

WANTED

Audio Engineering magazines prior to March of 1952. This is the old name for *Audio Magazine*. Call Mike, 708-431-4560, or e-mail EsotericTT@aol.com.

Looking to contact Chris Merren, formerly of Caldwell, N.J., who wrote me on the IPC AM-1026 theater amps and Eico ST-70s about 5–6 years ago; would like to finally respond to your fine letter. Ian Baren, chinacave@aol.com.

WEBSITE RESOURCES

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Book Review

Paul Wilbur Klipsch—The Life . . . The Legend

by Maureen Barrett and Michael Klementovich

Reviewed by Richard Honeycutt

(Rutledge Books, Inc., PO Box 315, Bethel, CT 06801, 1-800-2RUTLEDGE, info@rutledgebooks.com, $24.95 plus shipping and handling.)

If you were asked to name the single person who exerted a formative influence on the top loudspeaker engineers at JBL, Electro-Voice, Frazier, and literally dozens of other top manufacturers, that person would have to be Paul Klipsch. Fellow of the IEEE, AES, and ASA, recipient of the AES Silver Medal for innovative contributions to loudspeaker design and studies of acoustic distortion, inductee into the Audio Hall of Fame and the Engineering and Science Hall of Fame, Paul was a man whose work is unforgettable.

But to those who knew the man, his work is no more unforgettable than Paul himself. The authors of this first biography of Paul Klipsch are music lovers and audiophiles whom the man and his work have long inspired. Their goal in writing this book seems to have been to introduce Paul's essence to those who never had the privilege of knowing him.

A personal note: knowing that I have long been an admirer of the American philosopher William James, my wife gave me a biography of James some years ago. The biography was a real disappointment, because, even while covering the life of Professor James in great detail, it all but ignored his thought and his work, which were the reasons for my interest in him. This is not an uncommon failing of biographies.

Unlike the biographer of William James, Barrett and Klementovich took the high road, in my estimation, in writing their biography of Klipsch. They included discussions of his inventions, his values concerning audio in general and loudspeakers in particular, his eccentricity, his wit and wisdom–the essence of who Paul was. In this they are to be commended.

They added favorite stories about Paul taken from published and unpublished sources. They included a family gallery of the various models of speakers Paul designed. If you are hoping for long chapters filled with anecdotes from grade school, tear-jerkers about how hard a time Paul had financially in the early days of his company, and photos of Klipsch relatives to the n^{th} degree, you will be disappointed. But if, instead, you wish you knew Paul Klipsch, or you did know him and enjoyed his personality, you will find this book delightful.

Here's a taste of details you'll find in the biography: Paul's office–the OUT box read "OUT" and the IN box read "STAGNATING." Signs on the wall read: "The Lord Giveth and the Government taketh away," "My mind is made up; don't confuse me with facts," and "A clean uncluttered desk is the sign of a sick mind."

Another example is the story of how Paul bought a calculator and the insert said it wouldn't operate in high temperatures. Being intrigued, he took it back to the office and turned the temperature up extremely high. He stripped down to his skivvies and proceeded to dismantle the calculator and test it in various ways. He found the reason it wouldn't work at high temperatures and wrote the manufacturer to tell them why, and what to do about it.

If you have read between the lines and suspect that the Barrett/Klementovich biography engages in more than a bit of hero worship, you are correct. But their subject forces them to it. At lunch with a couple of Klipsch engineers some years ago, we discussed Don Davis, Don Keele, and other gurus who had been a part of the Klipsch cadre at one time or another. One of the engineers summed it up nicely: "We're all really just disciples of Paul Klipsch."

For us disciples of Paul Klipsch–and our name is Legion–this book is required reading. ❖