Crossover Tips from a Pro

audioXPRESS

the AUDIO TECHNOLOGY AUTHORITY

MARCH 2005
US $7.00 Canada $10.00

Back

Affordable, High-Quality **Starter Kits**

Small **French Speakers** Are Perfect Fit

Kit-Building **MTM Units**

A Minimalist **Look At** Tube Amp Design

A Simple Music Library Preamp

Buffered, Isolated RIAA Emphasis Network

www.audioXpress.com
03
0 74470 84697 0

solid state

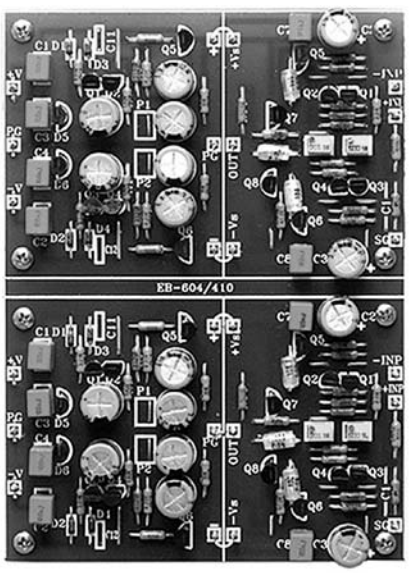
Starter Kits, Part I

EB-604/410 All-JFET Line Amp

By Erno Borbely

This noted circuit designer begins his starter kit series with an all-JFET line amp.

PHOTO 1: THE EB-604/410 LINE AMP WITH ON-BOARD REGULATORS.



If you have been following my designs through articles in Audio Amateur publications, you know that my priorities are lowest possible noise, best inherent linearity, and DC coupling. Low noise, of course, is a necessity in this digital age when you are playing CDs with >100dB S/N ratio, but to me a low-noise phono preamp and a low-noise microphone preamp are even more important. My interest in inherent linearity stems from my objective of not having to use negative feedback to "clean up" a bad design.

I use feedback primarily to stabilize certain DC or AC parameters of the amps or to achieve special frequency responses, such as in RIAA phono preamps or crossover filters. I use DC coupling whenever possible, because I don't like "capacitor sound" and also because I am "allergic" to the prices of so-called high-end capacitors. Of course, all of this should always go hand in hand with the overall objective in audio: good sound.

Having said that, I also must admit that these objectives can drive the prices to a level where only a handful of people can afford them. And yes, I do design such "price no object" custom amps for those who have extremely high-resolution systems. However, I would also like people with limited budgets to experience and enjoy my designs. Consequently, I have given up some of the highest design criteria and have come up with designs that are affordable, yet still offer high sound quality, carrying the Borbely Audio sound signature of high resolution and transparency! In this context I designed a series of new kits called Starter kits.

The Starter kits are relatively simple and inexpensive JFET or all-FET circuits. The "secret" of the high-quality sound lies not only in the topologies, but also in the passive components used. I have also included simple discrete all-FET regulators for each of the circuits, because I don't want you to use 3-pin LM317/337 type regulators, which are not up to the task required by today's audio amps (see the note on my home page under Upgrades).

These circuits are further developments of the basic ones described in previous articles ("JFETs: The New Frontiers, Part I and II," *AE* 5/99 and 6/99, and "All-FET Line Amp," *aX* 5/02.) You'll find copies of these articles on my homepage under Special Articles. Please print out and read these articles because I am going to refer to them frequently in the description of the new ones.

DEVELOPING THE NEW TOPOLOGY

The differential amplifier, which was described in Part II of the JFET articles (see Fig. 12A/B, p. 16, in *AE* 6/99), is the basic building block on which many—if not all—of my amplifiers are based (more on "if not all" in Part II).

I'll give a quick recap of the operation of the differential amplifier here, but you can find a more thorough description in Part II of the JFETs article. *Figure 1A* shows the basic configuration, which essentially joins two common emitter amplifiers by their sources. You can do this as long as the two JFETs are identical. If the I_0 is constant, then $I_{D1} + I_{D2}$ is also constant because $I_{D1} + I_{D2} = I_0$.

ABOUT THE AUTHOR

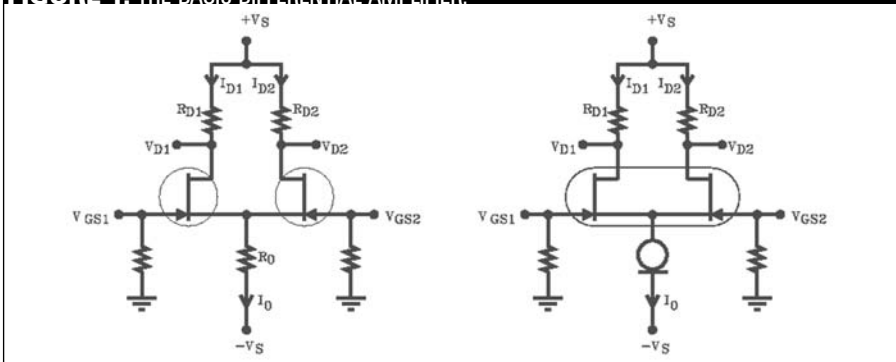
Erno Borbely received an MSc degree in Electronic Engineering from the Norwegian Institute of Technology, University of Trondheim. In the 60s he was working as a design engineer for the Norwegian Broadcasting Corp., developing studio equipment, both in tube and in semiconductor technology. In 1969 he moved to the US and worked for David Hafler at Dynaco. Mr. Borbely was named Director of Engineering for Dynaco in 1972. He developed the Cascode output stage for the Stereo 400 power amplifier, and the Dynatune circuit for the FM5 FM tuner, for which he received a US patent.

In the 70s he worked at Motorola Semiconductor, developing power amplifiers, low-noise preamplifiers, and FM tuners. Some of this work was put to good use, when he joined the David Hafler Company in 1978 as Director of Engineering. Mr. Borbely developed the DH101 preamplifier and the DH200 power amplifier, which was the first one to use MOSFETs in the US.

From 1980 until 1997 he worked for National Semiconductor in Germany as its Technical Training Manager. In 1984 Mr. Borbely and his wife Irene started Borbely Audio, which sells high-quality kits to end-users. Since 1982 he has been publishing his designs in the Audio Amateur publications (*Audio Amateur/Audio Electronics, Speaker Builder, Glass Audio,* and now in *audioXpress*). In the last several years Mr. Borbely has developed a line of all-FET audio amplifiers. Besides the kit business, Mr. Borbely is designing audio products for OEM customers.

Mr. Borbely's web page address is www.borbelyaudio.com and his e-mail address is erno@borbelyaudio.com.

FIGURE 1: THE BASIC DIFFERENTIAL AMPLIFIER.



The prerequisite for this is that R_0 is very large. For R_0 to be very large $-V_S$ must also be very large. This is usually inconvenient, and instead of making R_0 large, replace it with a constant-current source as shown in *Fig. 1B.*

The operation of the differential or balanced amplifier is simple. Holding V_{GS2} constant and increasing V_{GS1} will increase I_{D1}. However, $I_{D1} + I_{D2} = I_0$, so if I_{D1} increases, then I_{D2} must decrease. This will cause V_{D1} to decrease and V_{D2} to increase for a differential voltage at the output. If you change V_{GS1} and V_{GS2} at the same time, but in opposite directions, then you achieve the same effect, but even more of it—i.e., you have more differential gain. This is now a truly differential amplifier with gain of:

$$A_V(DD) = (V_{D1}–V_{D2})/(V_{GS1}–V_{GS2})$$

Now, what happens if you increase V_{GS1} and V_{GS2} in the same direction? If both are going positive, then both drain currents should increase. However, this is impossible, because $I_{D1} + I_{D2} = I_0$. Consequently, the differential amplifier cannot amplify same-polarity or common-mode signals.

Based on this differential amplifier, I designed a balanced/SE all-JFET line amp, described in Part II of the article (see Fig. 17 on p. 18; also available as a kit under the designation: EB-2000/402, see my homepage). Many customers, including pro audio people, built this amp and called it a "very musical" amplifier. The only problem I have with this circuit is that the outputs are not sitting at 0V DC and a capacitor is necessary to couple the output to the outside world. I generally don't like caps in the signal path, even if they are made of 99.99% silver. (I must admit that part of my dislike is caused by the fact that people are charging outrageous prices for the caps.)

Fortunately, it's easy to design DC-coupled amplifiers because of availability of complementary devices. *Figure 2* shows how a common-source amplifier used as a level shift device can turn the differential amplifier into a DC-coupled one.

DC COUPLING

Figure 2A shows the basic circuit. The input JFETs are closely matched and are operated at approximately 2mA each. The constant-current source that supplies the 4mA is made up of two J508s or two E-202s in parallel. The 2mA drain current in each JFET causes a voltage drop of 3V across each of the 1k5 resistors. This 3V produces about 3.1V across the 332R source resistor of the J74, setting up approximately 9mA in the second stage.

I tried the circuit first with a resistive drain load of 2k49 in the second stage. This resulted in an open loop gain (OLG) of only 27dB. The linearity was also less than adequate with an open loop THD of 0.2% at 3V/1kHz. When I replaced the 2k49 with a 10mA constant-current source, the OLG increased to approximately 40dB and the linearity also improved to 0.075% THD at 3V/1kHz.

The drain of the J74 sits at close to zero volts, but a certain offset trimming is necessary. The 50R trimpot serves this purpose. The 10k-1k1 feedback resistors give a closed loop gain (CLG) of 10× (20dB). The closed loop THD at 20dB gain is 0.0045% at 3V/1kHz and 0.02% at 3V/10kHz.

If you operate the amp with a CLG of 6dB, then you need to put a 22pF cap across the feedback resistor. The THD is reduced to 0.0017 and 0.0037%, respectively. The circuit is not unity gain stable (NUGS), but you can force it by dramatically increasing the cap across the feedback resistor to 220pF! Conclusion: don't use it as a unity gain buffer!

You can further improve the linearity by converting the common-source second stage to a folded cascode (*Fig. 2B*). Here the second stage is driven not only by the gate but also by the source. The second stage then becomes a part of the first stage, coupled to it through the drain resistors.

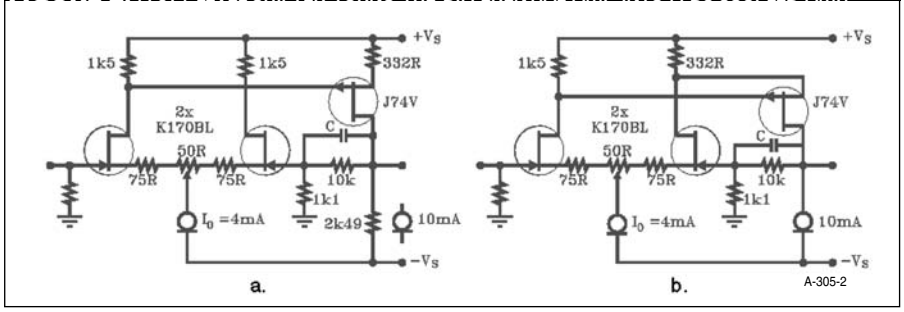
The 20dB CL THD is 0.0035% at 3V/1kHz and 0.013% at 3V/10kHz. The 6dB figures become <0.001 and 0.0017, respectively. The high frequencies benefit most from this conversion. The maximum output swing with ±24V supply is about 10V RMS. Minimum load is 10k. By the way, the amp is still NUGS!

You can also use these three-JFET amps in a unity gain inverting configuration. Connect the + input to ground, lift off the 1k1 from ground and use this as the –input. I tried it with 10k–10k for 0.001% THD at 3V/1kHz and 0.0032% at 3V/10kHz. This is a very simple I/V converter for your CD player.

ENTER COMPLEMENTARY SYMMETRY

The three-JFET circuit in *Fig. 2B* is essentially a simple discrete op amp, with very decent performance. However, in my experience, complementary symmetry improves the neutrality and transparency of the amp. This, I believe, comes

FIGURE 2: A J74 LEVEL-SHIFTER TURNS THE DIFFERENTIAL AMP INTO A DC-COUPLED AMP.



from further improvement of the inherent linearity, which manifests itself not so much in less measured THD, but in the way the distortion products look, indicating fewer higher frequency components.

I have therefore designed a fully complementary version of the simple amp shown in *Fig. 2B* (see *Fig. 3*). Consider the circuit in *Fig. 2B* as half of a complementary circuit. I have folded this down along the ground line, naturally replacing the n-channel JFETs with p and vice versa.

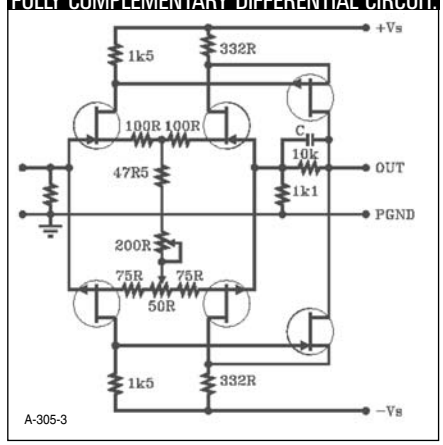
You can feed the input stages of this new circuit with 4mA constant-current diodes as before, but because the sources of the upper JFETs are sitting at a small but positive voltage and the lower ones at a negative voltage, they can be joined together through a resistor and will feed each other with the necessary source current. A simple trimpot will allow you to adjust this current to the desired level. An offset adjustment is also necessary as before.

The second-stage JFETs are connected together at their drains, acting as constant-current sources to each other. The whole amp is now a transconductance amplifier, converting input voltage to output current. You convert this output current back to voltage by connecting a resistor from output to ground. Here this resistor is connected back to the negative input of the differential input stage, acting both as a load on the output and a feedback resistor.

With the same feedback networks as before (10k-1k1), the OLG has increased to 300×, which is almost 50dB. The OL THD at 3V went down to 0.06%. The 20dB THD figures are 0.002 and 0.0059%, which are almost 6dB better than the three-JFET circuit.

Interestingly enough, the transconductance amp can drive just about anything within its current capability, without much change in the magnitude and structure of the THD. This is an advantage, because a simple load resistor can change the OLG and the OL frequency response of the amplifier without affecting the linearity. Nevertheless, I prefer to isolate the transconductance amp from the outside world with an output buffer, partially to allow low impedance loads, but also to control the OLG independently of the load. This will be helpful in an RIAA phono preamp, as you will see later.

FIGURE 3: THE CIRCUIT CONVERTED TO A FULLY COMPLEMENTARY DIFFERENTIAL CIRCUIT.



THE EB-604/410 LINE AMP

Based on the transconductance circuit developed in *Fig. 3*, I designed a complete line amp. The Starter kit EB-604/410 line amp is a high-quality, low-noise, low-cost all-JFET circuit, with on-board regulators. It is designed to work with 6–20dB gain. For gains up to 12dB, you can use single-matched JFETs (delivered with the kits).

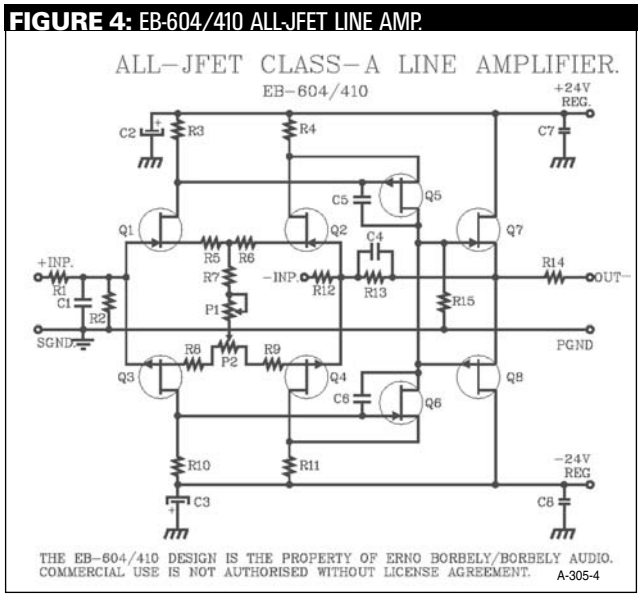
For DC stability, I recommend dual monolithic JFETs at 20dB gain. The PCB is 105 × 145mm and contains two completely independent circuits. The amps can be supplied from one common or two independent power supplies. The schematic is shown in *Fig. 4*.

The input stage is a complementary differential circuit, consisting of Q1–Q4. Each of these JFETs is working with 2mA drain current. Q1–Q2 (K170BL/V) and Q3–Q4 (J74BL/V) are closely matched devices. Alternatively, you can use dual monolithic JFETs for Q1–Q2 and Q3–Q4.

P1 is setting up the drain current to 2mA and P2 is used to adjust the output offset to 0V. The second stage Q5–Q6 (J74V/K170V) is a folded cascode, coupled to the input stage through resistors R3/R4 and R10/R11. It is using matched high Idss JFETs, and is operating at close to 10mA. John Curl used the same folded cascode configuration in his JC-2 phono preamp module, but he used bipolar transistors for the folded cascode.

The output stage, consisting of Q7–Q8, is a complementary JFET follower circuit (see description in Part II, p. 18, Fig. 16B, of the JFETs article). Again, these are closely matched devices, operating at <9mA, to limit their power dissipation. If you use higher Idss devices, you must put small TO-92 heatsinks on the devices. The circuit is not unity gain stable without the caps C5 and C6 (100pF). At gains >6dB you can reduce the caps to 33pF.

FIGURE 4: EB-604/410 ALL-JFET LINE AMP.



You will recall from the JFET articles that the K170 and J74 JFETs have relatively high input capacitance. Unfortunately, this input capacitance is also voltage dependent, which might cause distortion in certain cases. This shows up mostly when you drive the circuit from high source impedance. I checked this for the 410 and found that there is no change up to a source impedance of 5kΩ. This is equivalent to a volume control of 20kΩ.

If you need higher value volume control, you can use JFETs with lower input capacitance for Q1/Q2 and Q3/Q4. The high V_p JFETs have significantly lower input capacitance. I have tried the K246BL and J103BL for Q1/Q2 and Q3/Q4, respectively, and they work fine. All you need to change is the resistor connecting the two sources together. Since the gate-source voltage is much higher for the high V_p devices, the resistor must be bigger to set up the same drain current as before. For the K246BL/J103BL it's about 560R, so use a 332R in series with a 500R trimpot. With the K246BBL/J103BL you can use a 100k volume control without any change in distortion.

Why not use the K246BL/J103BL in the first place? The high V_p devices have higher noise than the low V_p ones, and also the inherent linearity is better with the low V_p devices. Nevertheless, the high V_p devices offer a very good compromise between low input capaci-

AFTER 20 YEARS, BORBELY AUDIO IS STILL GOING STRONG

This year marks the 20th anniversary of Borbely Audio, and, although the DIY audio enthusiasts have changed over the years, they continue to be the driving force of our business. Needless to say, our years of experience studying our customers has given us a wealth of information about who they are, why they choose our kits, and what more contributions can Borbely Audio make to the audio market.

To give you a better understanding of how we've determined these, I invite you to view our new online system at www.borbelyaudio.com. This year Borbely Audio took advantage of technology and transferred all kit documentation, ordering and fulfillment to an online process that enables us to cover and segment our kit offerings from the High-Performance kits, Pro-Audio kits, and our newly released "Starter Kits." Over the past year or so, we've received numerous e-mails, letters, and other correspondence from customers requesting additional kits that are tailored for those beginning their quest for high performance audio.

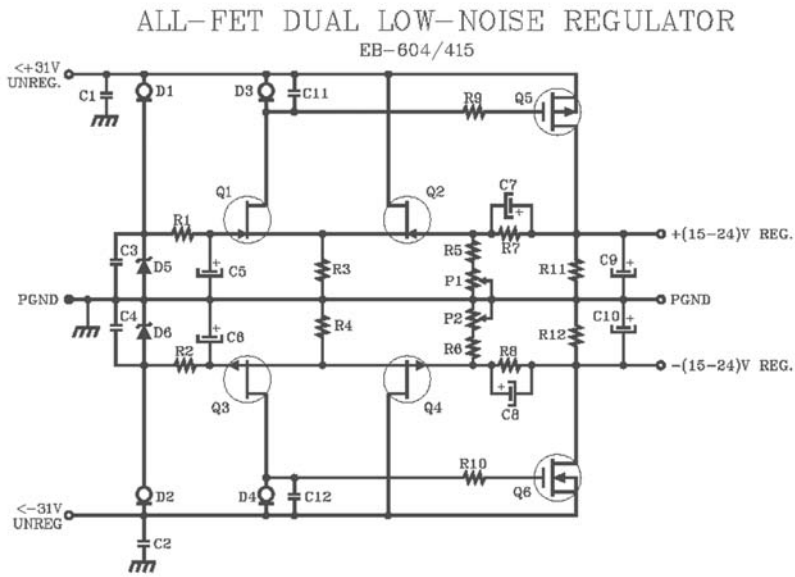
To that end, Borbely Audio is very proud to release new kits designed to ensure maximum reliability and performance for the new kit builder. The "Starter Kits" are based on complementary differential designs, exacting superior performance with the same superior components as our other kits share. All of the Starter Kits have onboard regulators and newly designed power supplies to complement the kit. We've been able to design these new kits more reasonably priced and yet continue to use the superior components you've come to expect. DIYers everywhere have learned that not all components are made equal.

Our goal as always is to obtain the best possible performance from our components. This is a time-consuming yet worthy task. Components found in many "chain" electronic stores are far inferior in build quality and tolerance. This is one of the single most important aspects for kit builders everywhere. Price is not as strong a factor as you might think with today's DIYers, as we've found price was never a primary issue when exceptional performance was involved. In most cases, price came in fourth behind essentials such as "quality," "availability," and "technical support." The new "Starter Kits" provide the best of both price and performance. During the next few months we'll take an in-depth look at these new kits in the pages of *audioXpress*.

One final observation before turning you loose on the new kits—it's obvious that today's DIY audio kit builders are still around in force. If there is a new design or kit that can be found, they'll find it, simply because they love the experience. The number of online forums for kit builders and DIY audio websites indicates that DIYers are not going anywhere, despite what some might have you believe. And as long as they are interested in building superior audio equipment, we'll be around to offer it to them.

—**Larry Black, Audiokits.com**

FIGURE 5: EB-604/415 ALL-FET REGULATOR ON THE EB-604/410 BOARD.



tance and noise/linearity. I have also tried other high V_p devices, such as the 2N5458 N-channel and the 2N5461 P-channel devices, which also work satisfactorily. Don't hesitate to experiment a bit; you have a good chance of getting a very good result.

Only high-quality components are used in the amplifiers. The resistors are non-magnetic DALE CMF–55–143, the electrolytic caps are Nichicon FG-type "Fine Gold" MUSE, and the rest are polypropylene and polystyrene caps. The PCB is FR-4, with the component side printed.

Typical specifications with 6dB gain:

Power consumption per channel: 25mA
Open loop gain: 58dB
Open loop frequency response: 20kHz (–3dB, C5/C6 open)
Closed loop frequency response: >1MHz
Closed Loop THD: 3V/1kHz: 0.0015%
3V/10kHz: 0.0025%
Equiv. shorted input noise: approximately 0.3µV*
Input capacitance: with K170/J74: 45pF
with K246/J103: 7.5pF
with 2N5458/2N5462: 4.5pF
Output impedance: <1Ω (approximately 50Ω with R14)
Minimum load: 1kΩ
*Measured with R12 = 10R, otherwise the noise is dominated by resistor noise.

ON-BOARD REGULATOR

The schematic of the discrete, on-board all-FET regulator is shown in *Fig. 5.*

The regulators are discrete FET op amps; the positive regulator is made up of differential JFET pair Q1–Q2 (K170BL/V) and MOSFET Q5 (J148), the negative of Q3–Q4 (J74BL/V), and Q6 (K982). Q1–Q2 and Q3–Q4 are closely matched devices. Due to their relatively low breakdown voltage, the maximum input voltage to the regulators is limited to 31V. If the input voltage is >31V the higher-voltage dual JFETs must be used. The differential pair JFETs are working

FIGURE 6: STUFFING GUIDE FOR EB-604/410 WITH ON-BOARD REGULATORS.

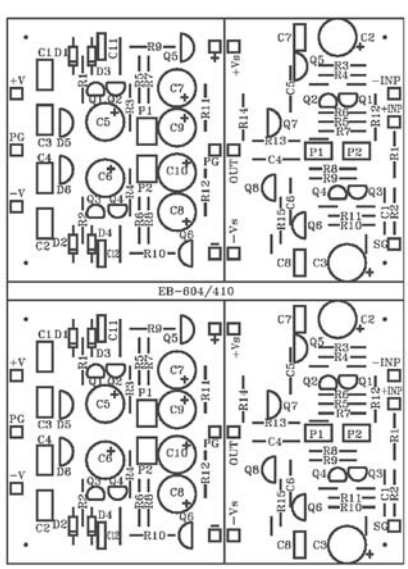


FIGURE 7: EB-604/410 USED AS AN UNBALANCED, NON-INVERTED LINE AMP.

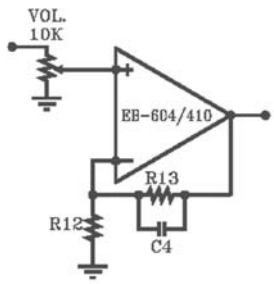
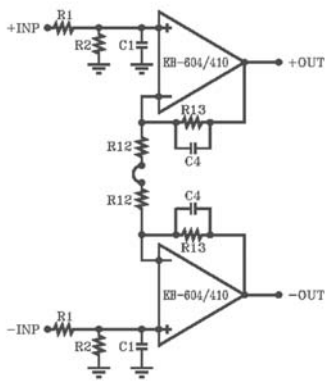


FIGURE 8: TWO EB-604/410 USED IN BALANCED MODE.



at 2mA drain current each, which is set up through the 6.9V reference diodes D5 and D6 and source resistors R3 and R4.

The reference diodes are fed from constant-current diodes D1 and D2. The output MOSFETs Q5 and Q6 are TO-92 devices and are capable of supplying maximum 35mA. The minimum input-output voltage difference is 2V, but I recommend 4V. If the output voltage is fixed at 24V, then adjustment is not necessary, and P1 and P2 can be left out (R5 = R6 = 3k32, P1 and P2 are shorted). The output noise of the regulators is <5µV across the audio bandwidth.

Stuffing guide for the EB-604/410 line amp with on-board regulators is shown in *Fig. 6*. Recommended power supply for the 410/415 combination is the EB-604/263.

SETUP PROCEDURE

If possible, test each amplifier module separately before installing it in the chassis. This simplifies measurements, adjustments, and if necessary, component changes. You need only a basic digital voltmeter (DVM) for the adjustments. However, if you have access to a scope, connect it to the output of the line amp and check whether radio frequency (RF) oscillations are present. If you have complete audio instrumentation in your workshop, perform the usual gain, frequency response, noise, total harmonic distortion (THD), and intermodulation distortion (IM) measurements.

FIGURE 9: WIRING DIAGRAM FOR THE EB-604/410 LINE AMP WITH THE EB-604/263 POWER SUPPLY.

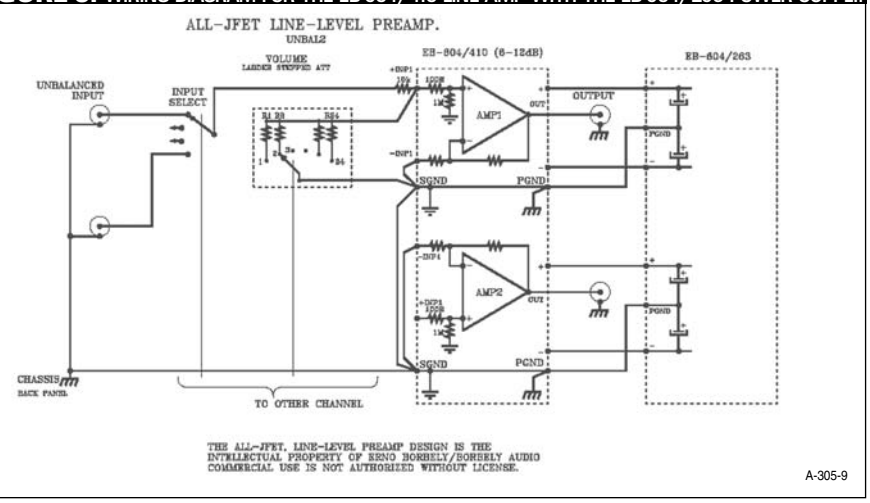
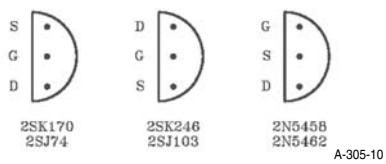


FIGURE 10: PINOUT FOR THE DIFFERENT TYPES OF JFETS (BOTTOM VIEW).



Start the assembly with the regulators. Insert the solder pins, then add the resistors and the FETs. Finally, install the caps. Connect a 1kΩ/1W resistor between the + output and ground and the – output and ground. Connect ±29V unregulated DC to the regulator inputs and check the regulated voltage with the DVM. The output voltage should be ±24V, ±5%.

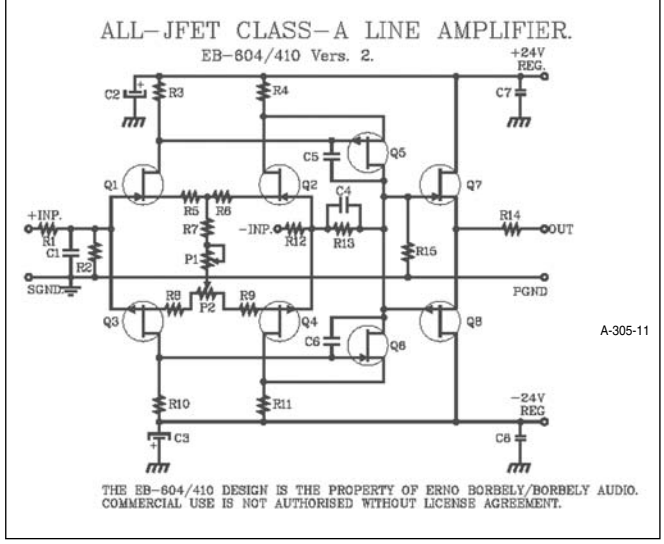
Next, assemble the amps. Start with the jumpers, then solder the resistors and the JFETs. Finally, add the caps. Connect the supply jumpers between the regulator and the amp. Connect the +INP of the amps to the input ground. Apply the unregulated supply voltage to the regulators. Check the voltage drop across R4 (or R11) with the DVM. Adjust the voltage drop to 3V with P1. Connect the DVM to the output of the amp and check the offset. Adjust the offset to 0V with P2. The amp is now ready to be installed into the chassis.

APPLICATION NOTES

For 6dB gain connect –INP to ground (R12 = R13 = 2k21). For 3dB gain R12 is 5.6k. If you need 12dB gain, change R12 to 732R or 750R. For a normal, unbalanced, non-inverting line amp, use a 10k or 20k volume control in front of it (*Fig. 7*). Because of capacitance modulation, I do not recommend volume control higher than 20k, but rather the ALPS 27mm Blue or the 40mm Black potentiometer for volume control.

You can also use two of the EB-604/410s in balanced operation, or as unbalanced-to-balanced converter and

FIGURE 11: THE OUTPUT STAGE IS NOT INCLUDED IN THE GLOBAL FEEDBACK LOOP.



balanced-to-unbalanced converter (*Fig. 8*). For balanced mode, connect the two –INP on the boards together. When used as an unbalanced-to-balanced converter, connect the +INP of the lower amp to ground. For balanced-to-unbalanced operation use the output of the upper amp as the output.

Figure 9 shows the wiring of the line amp. The vertical ground wire on the left-hand side is a bus-bar connecting the ground side of all the RCA connectors together. This is grounded to the chassis, normally on the back panel. You then connect a single insulated wire from this point to the SGND on amp 2. If the power supply is in the same box as the amp then I recommend shielded wire from the RCAs to the input selector switch and from the switch to the inputs.

PARTS LIST

EB-604/410

RESISTORS, TRIMPOTS

R1, R5, R6	100R
R2	1M
R3, R10	1k5
R4, R11	332R
R7, R14	47R5
R8, R9	75R
R12, R13	2k21
R15	47k5
P1	200R COPAL
P2	50R COPAL

CAPACITORS

C1, C5, C6	100pF, PS
C2, C3	47µF, 35V (LS: 5mm)
C4	100pF, PS
C7, C8	0.1µF, PP (LS: 5mm)

FETS, DIODES

Q1, Q2	K170BL/V (matched to Q3, Q4)
Q3, Q4	J74BL/V (matched to Q1, Q2)
Q5	J74V (Idss: >10mA, matched to Q6)
Q6	K170V (Idss: >10mA, matched to Q5)
Q7	K170BL (Idss: 8–10mA, matched to Q8)
Q8	J74BL (Idss: 8–10mA, matched to Q7)

MISCELLANEOUS

24× 1mm solder pins
PCB: EB-604/410

EB-604/415

RESISTORS, TRIMPOTS

R1, R2, R5, R6	3k09
R3, R4	1k82
R7, R8	8k25
R9, R10	100R
R11, R12	10k
P1, P2	5k COPAL

CAPACITORS

C1, C2, C3, C4	0.1µF, 100V, PP
C5, C6	220µF, 16V
C7, C8, C9, C10	47µF, 35V
C11, C12	Not used

FETS, DIODES

Q1, Q2	K170BL/V (matched, Idss: >9mA)
Q3, Q4	J74BL/V (matched, Idss: >9mA)
Q5	J148
Q6	K982
D1, D2	E202 (2mA)
D3, D4	E102 (1mA)
D5, D6	LM329DZ (6.9V ref.)

EXPERIMENTERS AHOY!

Part of my intention in writing this article was to give you a chance to experiment with the circuits yourself. I mentioned that you could use high V_p devices in the input stage, which gives lower input capacitance. I have not tried high V_p devices for the second stage, but you could try some high Idss ones here, too. The K246 and J103 do not come with high enough Idss, so you must find some others.

On the other hand you can use the K246 and the J103 for the complementary source-follower output. Select the highest Idss devices from the "BL" group and match them to ±10% of Idss. You will lose some inherent linearity using the K246BL/J103BL devices, but they are nevertheless good alternatives for the higher priced K170/J74. Note that the pinout is different for the three types of JFETs (*Fig. 10*).

Some readers have asked whether you can use the circuit without "global feedback." In this case the global feedback means whether or not the output stage is included in the feedback loop. *Figure 11* shows the 410 schematic in which the feedback is taken from the second stage. Compare this circuit with *Fig. 3* to see that the input and the second stage are the same as in *Fig. 3*, with the addition of a complementary source follower stage. Some people believe it should be used this way because the amp sounds softer. I suggest you try it both ways, and choose the configuration that best fits your listening preferences.

The K170BL/J74BL complementary JFET source-follower output stage can drive loads down to 1kΩ. Should you need to drive a 600Ω Sennheiser headphone, you can use two pairs of K170BL/J74BL in parallel. Some customers have used four pairs of these in parallel to drive 300Ω headphones!

Naturally you can also elevate this design into a higher category by using some of the upgrades I described in the Upgrades notes on the homepage. The Dale CMF-55 143 non-magnetic resistors are very good sounding, but using Caddocks MK-132 instead gives you even more resolution and dynamics. The most dramatic improvement would come from using a stepped attenuator at the input. Try the fixed-series/variable shunt attenuator (*Fig. 9*), and you are getting close to my high-performance kits. ***aX***

SOURCES

AUDIOKITS.COM
36 Applebee Farms Dr.
Jacksonville, Ill. 62650
(217) 243-1006
(217) 245-5694 (Fax)
1-888-863-3482 (Toll-free)
sales@audiokits.com
www.audiokits.com

ACKNOWLEDGEMENTS

My sincere thanks to Walt Jung, who kindly read the manuscript and provided valuable comments and suggestions.

Also thanks to North American distributor Audiokits .com for their continued support and help in the Starter kits development program

All-JFET MM/MC Phono Preamp with regulators — Starter Kits

Lt.	Type nr.	Description	Comments	Price
1	EB-804/419	All-JFET MM/MC Preamp w/ regulator	FR-4 PCB, Dale Resistors	$349.80
2	EB-604/265*	2×10VA 22V Power Supply	FR-4 PCB	$151.80

* Must be mounted in a separate case

All-JFET I/V Converter and Filter for DACs with regulators

Lt.	Type nr.	Description	Comments	Price
1	EB-604/417	All-FET I/V Converter and Filter for DACS w/ regulator	FR-4 PCB, Dale Resistors	$250.80
2	EB-604/263	2×7VA/18V Power Supply	FR-4 PCB	$130.68

All-JFET line amp with regulators

Lt.	Type nr.	Description	Comments	Price
1	EB-604/410	All-JFET line amp with regulator	FR-4 PCB, Dale Resistors	$211.20
2	EB-604/263	2×7VA/18V AC Power Supply	FR-4 PCB	$130.68

Tubes

A Minimalist Amplifier Design

Back

By Larry Lisle

If you're tired of the noise and distortion of resistors and capacitors in the signal path, try the minimalist approach to amplifier design and experience something new!

PHOTO 1: A "MINIMALIST" AMPLIFIER WITH ONLY THREE PARTS. THERE ARE NO RESISTORS OR CAPACITORS IN THE SIGNAL PATH, ONLY TWO EXCELLENT TRANSFORMERS AND A GREAT TUBE. IT'S LIKE OPENING A DOOR TO THE MUSIC!

From the beginning of electronic audio reproduction, the major design trends have been to achieve wider bandwidth, lower distortion, and greater power. An unwanted byproduct has usually been more complicated equipment with more and more parts. Unfortunately, each part can add its own coloration to the sound. It may not be readily measurable but it is perceivable, especially in a long listening session.

The minimalist approach goes in the opposite direction. What can be done with the fewest components possible? One result has been an amplifier with three parts: an input transformer, a tube, and an output transformer.

The circuit (*Fig. 1*) is really from the 1920s, but they didn't have the music sources, the transformers, or the speakers to hear what it could do. With top-quality components, this amplifier is nearly flat from 20Hz to 20kHz, has low distortion, will provide a good listening level with sensitive speakers or earphones, is extremely quiet, and sounds absolutely gorgeous!

QUALITY, NOT QUANTITY

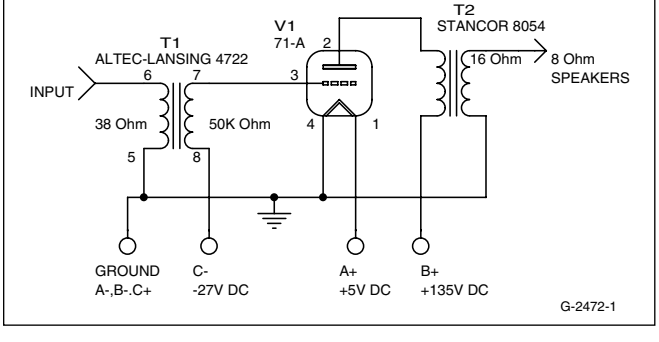
Few parts, however, does not mean cheap. You must use the highest quality parts in a minimalist design. There is no other circuitry to compensate; you must get the sound right the first time! The power supplies for the plate, grid, and filament also must be top quality.

This is not really a straightforward construction article because the parts used in the amplifier (*Photo 1*) haven't been made for many years. But you can find them—or near equivalents from the golden age—if you look hard enough. There are also modern manufactured substitutes that sound very nice. Remember: for stereo, you will need two of everything for the amplifiers. Also, I recommend two completely separate power supplies.

The amplifier shown is designed to take its input from the headphone jack of a CD player. The signal is fed into what is called an "input transformer" of the type used to match a dynamic microphone, voice coil, or line to a grid. The impedance of the input winding should approximately match that of the output of the CD player while the secondary for the grid is in the 40,000 to 80,000Ω range.

If you don't know the impedance of your player, you can try a value of about 50–150Ω, or you can find the exact value by the following procedure: Play a CD with a steady sine-wave tone and measure the open circuit voltage. Then place a variable resistor across the output. The value of resistance that reduces the output by half is the output impedance.

FIGURE 1: THE CIRCUIT OF THE MINIMALIST AMPLIFIER AS SHOWN IN THE PHOTO. IT'S A SIMPLE CIRCUIT, BUT IT'S NOT CHEAP! SEE THE TEXT FOR DETAILS.



By the way, many older input transformers were designed for speech only and emphasize the frequencies below 3kHz. You definitely don't want one of these for music. Before you pay big bucks for a transformer, be sure you know what you're getting!

PARTS SELECTION

Of the many transformers I auditioned for this article, my favorites were the Altec-Lansing Peerless transformers and those from United Transformer Company (UTC). Among modern manufacturers, I like the Hammond 806 and 808—which are available on special order from suppliers of Hammond products—and those from Electra-Print; the latter being custom-made to your specifications. Remember to stay below the maximum power level of the transformer. Amplifier performance with sample transformers is shown in *Fig. 2*.

For example, the Hammond 808 is rated at +15dBm. This means that in the 50Ω input configuration the input voltage should be less than 1.257V. See the *Radiotron Designer's Handbook*, Chapter 5, Section 3 (iii) (b) for more information on input transformers. The key differences in input transformers—separating the best from the good—are bandwidth in the circuit you're using and the production of odd harmonics in the transformer.

The tube I chose is the 71-A. I used to think nothing could beat a single-ended 45. I was wrong. A minimalist amplifier really shows what this beautiful tube can do. The 71-A (or older 171-A) is about the largest tube that you can, as a practical matter, light its filament with batteries.

Remember that I said a minimalist amplifier wasn't cheap! I use four alkaline D cells and a 25Ω variable resistor (*Fig. 3*). When the voltage drops below 5V with the resistor at minimum, I switch in a fifth battery. This scheme gives remarkably long life. You could, of course, use rechargeable batteries. You could use a conventional line-voltage-operated power supply, but it may not sound the same. Try batteries first.

OPERATION

As long as we're talking about power supplies, the high voltage of 135V is supplied by a transformer and a conventional 5U4 tube rectifier in a full-wave circuit (*Photo 3*). A separate transformer supplies 5V for the filament of the 5U4 and the positive voltage is taken from the center tap of the filament winding. The voltage then goes to a 2kΩ variable resistor to adjust the output to precisely 135V.

I did this to audition different parts under identical conditions in spite of changes in the line voltage. In usual practice you could replace the variable with a fixed resistor, which also serves to limit the sudden rush of current when the power supply is first turned on. This isn't really necessary as the 5U4 just loafs in this circuit.

Filtering is done by an electrolytic capacitor, a choke, an electrolytic capacitor, a choke, and paralleled 10 and 30μF oil-filled output capacitors. (Only the 10μF is shown connected in the photo.) Oil caps are big! I like the sound of an oil-filled capacitor in this position because it also serves as a plate return for the amplifier. A bleeder resistor is located across the output capacitor.

Be careful of all capacitors. Make sure they are discharged before touch-

PHOTO 2: A VARIETY OF INPUT TRANSFORMERS BY UTC, HAMMOND, ELECTRA-PRINT, AND ALTEC-LANSING.

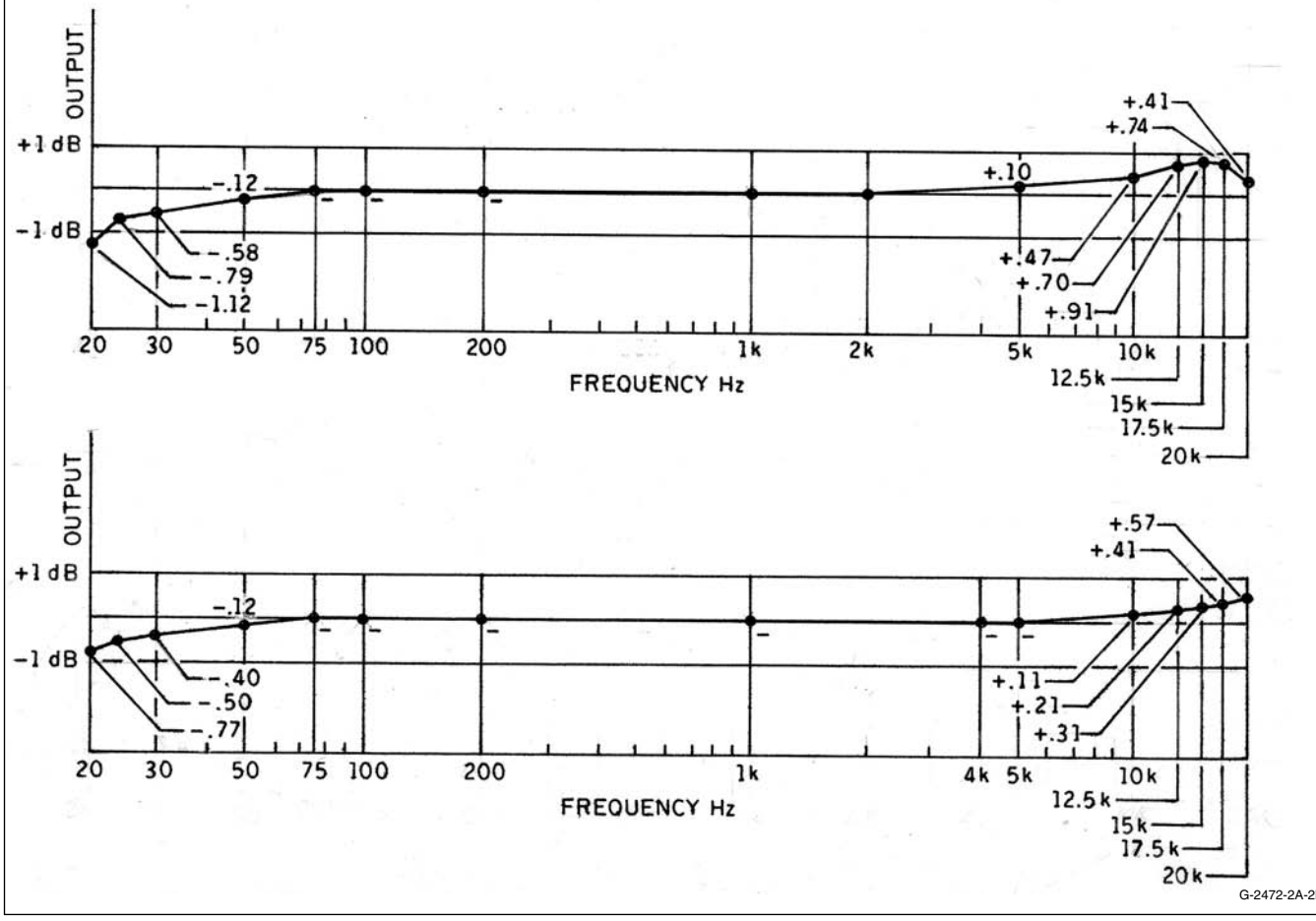
ing anything on the power supply or amplifier after unplugging from the AC outlet. Oil-filled capacitors especially can hold a charge for a long time. Don't trust the bleeder resistor. Discharge the capacitors manually.

Bias voltage is supplied by three 9V transistor batteries in series. The output transformer is a Stancor 8054 from the '50s. This is a push-pull transformer, but with the plate current of only about 14mA it shows no sign of saturation. It's one of the best-sounding transformers I have. The primary is 9kΩ, and I connect an 8Ω speaker load to the 16Ω tap of the secondary. This gives a load of 4500Ω and uses the entire secondary winding.

There are many single-ended output transformers on the market today, so finding something suitable shouldn't be a problem. If you happen to have a very high quality push-pull transformer though, give it a try. At low plate current many classics will actually sound and measure better than transformers designed for single-ended use.

Construction of the amplifier, the B supply and the A and C battery supplies (not shown in the photos) is on pieces of poplar 11½ × 7¼″ (actual measure). Assemble these in a wooden cabinet. (Call it 1920s surface mount.) I did this for sonic reasons, as I've found there are fewer problems with unwanted couplings using wood than with metal. If the transformers you're using aren't shielded properly, you may need to reconsider this point. Wires carrying A, B, and C voltages to the amplifier

FIGURE 2A, B: WITH GOOD TRANSFORMERS THE MINIMALIST AMPLIFIER IS ALMOST FLAT.



should be shielded and their shields connected to the amplifier's common ground point. Power supplies with AC inputs should not be located close to the amplifier.

CRITIQUE

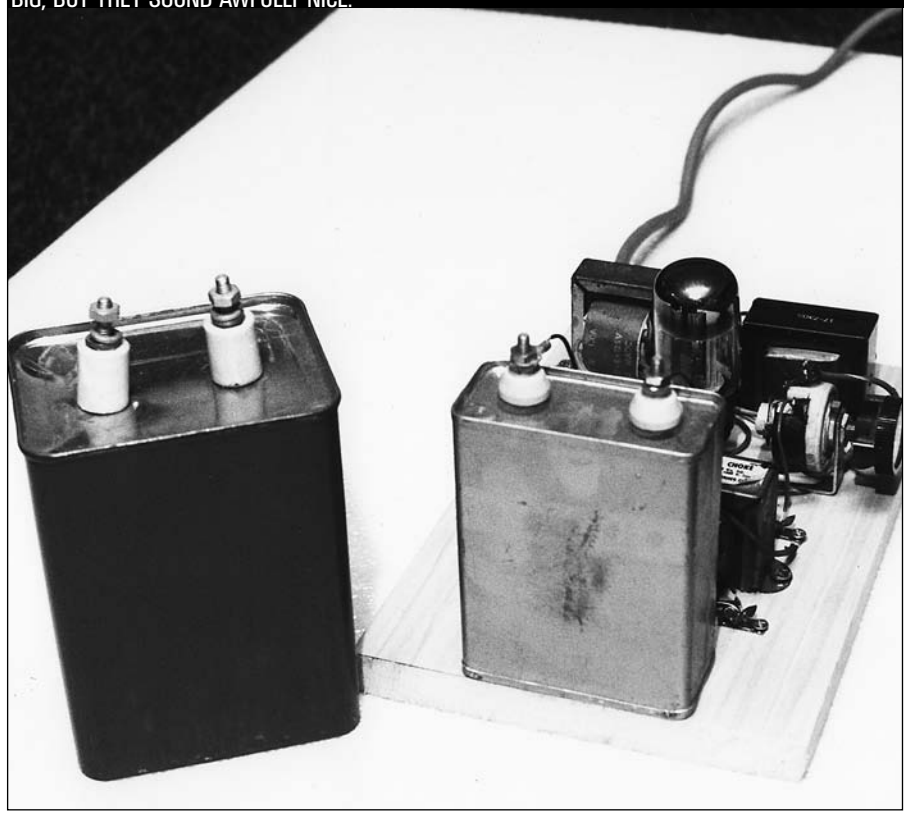
So how does it sound? Calling this amplifier "musical" is like calling Napoleon a soldier. It is a joy to listen to—it's like opening a door. It is extremely accurate. What goes in is what comes out, but louder. It's very close to the mythical "straight wire with gain" and a taste of single-ended goodness thrown in.

One of the most telling evaluations of an amplifier is how well it reproduces the human voice. We've had our brains wired since infancy to recognize voices and we're good at it. Few amps give the experience of a listener jerking his/her head around toward the speaker to see who is there. This amplifier does. It makes it seem real.

Sinatra on his best recordings is unbelievable. So is Ella Fitzgerald. The separation of instruments on Buddy Rich's "Big Swing Face" is outstanding. I'm hearing things on familiar recordings that I never noticed before. I'm also hearing deficiencies on some CDs that I had overlooked.

I didn't expect it to have much bass until I put on Bill Doggett's "All His Hits" or recordings by Sonny Burgess (maybe the best of the rockabilly artists), who knows what drums are for. Of course, it won't ripple the carpet, but it will reproduce what's on the CD at a comfortable listening level with sensitive speakers—no bricks with binding posts, please. It is a truly wonderful little amplifier. ***aX***

PHOTO 3: THE HIGH VOLTAGE, OR "B" POWER SUPPLY USES A 5U4 IN A FULL-WAVE CIRCUIT WITH PLENTY OF FILTERING. PARALLEL 10µF AND 30µF OIL-FILLED CAPACITORS ARE USED AT THE OUTPUT, WHICH ALSO SERVES AS THE PLATE RETURN CIRCUIT OF THE AMPLIFIER. OIL CAPS ARE BIG, BUT THEY SOUND AWFULLY NICE.

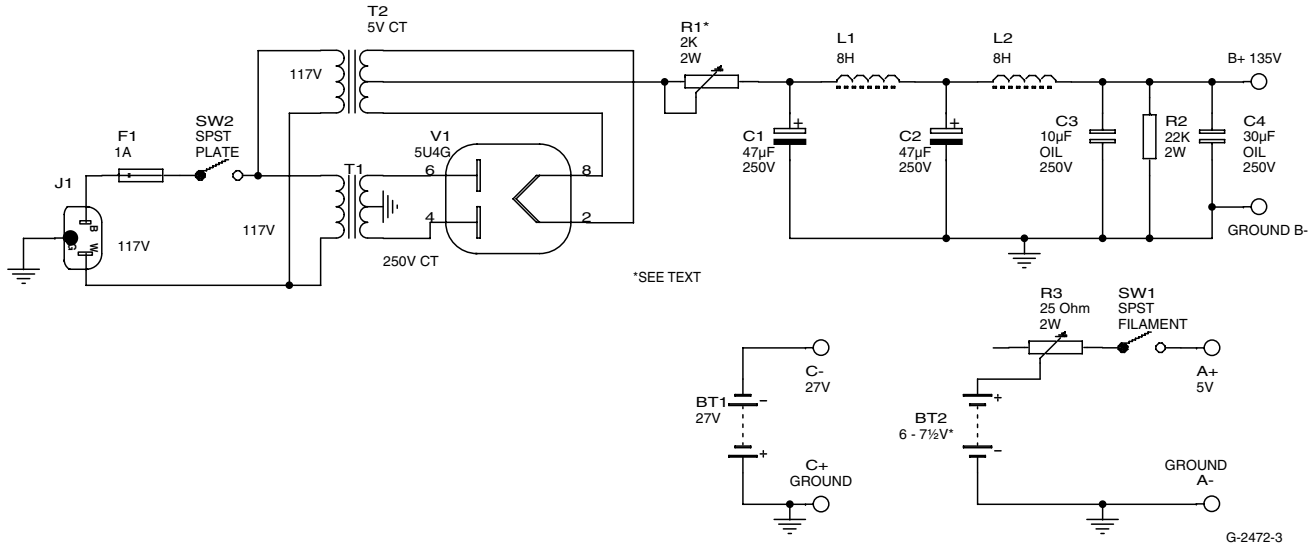


POWER SUPPLY PARTS LIST

BT1	27V (3, 9V batteries in series)
BT2	6 or 7½V (4 or 5 D cells in series—see text)
C1, C2	47µF × 250V electrolytic capacitors
C3	10µF oil capacitor, 250V or greater
C4	30µF oil capacitor, 250V or greater
L1, L2	8H or greater choke, 25mA or greater
F1	1A fast fuse
R1	2kΩ, 2W linear potentiometer
R2	22kΩ, 2W power
R3	25Ω, 2W linear potentiometer
SW1	Filament switch
SW2	Plate switch
T1	Mains to 250V center-tapped, 25mA or greater
T2	Mains to 5V center-tapped, 3A or greater
V1	5U4 full-wave vacuum rectifier (you may substitute others but it may require wiring changes).

Miscellaneous: cabinet, socket, knobs, wire, and so on.

FIGURE 3: A SUGGESTED POWER SUPPLY FOR THE MINIMALIST AMPLIFIER. TWO ARE NEEDED FOR STEREO.



speakers

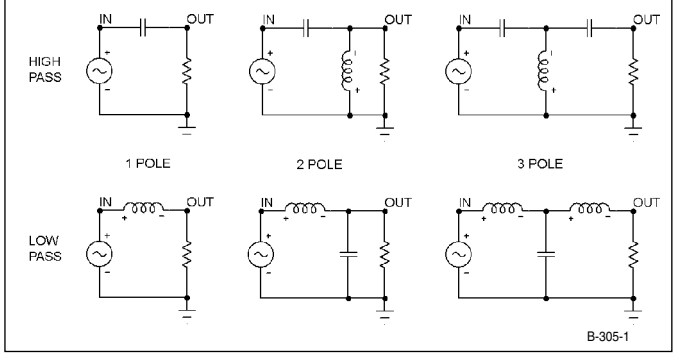
Current Source Crossover Filters

Back

By Nelson Pass

Tips from a pro about designing filter networks—for both voltage and current sources.

FIGURE 1: CLASSIC VOLTAGE SOURCE FILTERS.



In today's marketplace, audio power amplifiers are conventionally viewed as voltage sources, delivering a given voltage at the output that is a multiple of an input voltage. To the extent that they are truly voltage sources, having a very low output impedance, they simply deliver whatever current happens to reflect the response of the loudspeaker to the defined output voltage.

I have recently been playing with current source power amplifiers that have high output impedances and deliver a specific current to the load in response to an input voltage. The voltage across the loudspeaker reflects its response to this defined current. You could say that we have swapped the words voltage and current.

In a previous article, "Current Source Amps and Sensitive/Full-Range Drivers" (*aX* 10/04), I discussed some advantages of these amplifiers for Lowther, Fostex, and similar loudspeakers. In these cases the loudspeaker driver was "full-range," so there was no need for a crossover network to apportion the frequency spectrum between different drivers. However, I have found that these drivers typically appreciated a finite source impedance and often benefit from some equalization in the upper midrange and high frequencies.

With a voltage source, amplifier equalization is usually applied by a network in series with the driver. With a current source it is usually applied by elements placed in parallel, leaving the driver directly connected to the output terminals of the amplifier.

Of course, we don't want to limit ourselves strictly to one-driver loudspeaker systems (attractive as they may be), so we need to consider a means of filtering the signal to divide it between the drivers. We certainly can make life easy by using an electronic crossover network that does the filtering before multiple amplifiers. This way each driver gets an amplifier to itself. This has real performance advantages, but is generally more complex and expensive than a passive crossover between the amplifier and speaker.

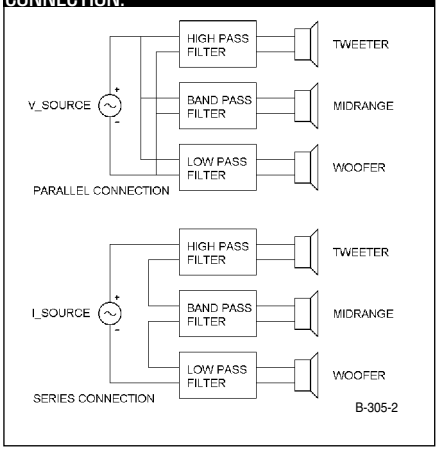
VOLTAGE SOURCE NETWORKS

The classic crossover networks of *Fig. 1* work very well if you have a voltage source. Here you see both high-pass (tweeter) and low-pass (woofer) filters with a resistor representing the loudspeaker. Of course, a real loudspeaker is not exactly a resistor, but I don't want to get confusing here. A low-pass network gradually attenuates the high frequencies as the frequency goes higher than a particular transition point called the crossover frequency. Similarly, a high-pass network gradually rolls off the frequencies lower than the crossover frequency. The shapes of both types of filters can be characterized by numbers in addition to the crossover frequency, but here we will limit ourselves to discussing just the slope.

The examples of *Fig. 1* show three different filters for both the high-pass and the low-pass filters. The first is a one-pole filter, corresponding to a rolloff slope of 6dB/octave. The second is a two-pole filter, which rolls off at 12dB/octave. The third is a three-pole filter, which rolls off at 18dB/octave.

There are, of course, filters with even more poles, and the exact shape of these filter curves can vary from sharply peaked at the "knee" of the crossover frequency to very gentle transitions. They all have different phase characteristics as well. There is no such thing as an ideal crossover network for all loud-

FIGURE 2: PARALLEL AND SERIES FILTER CONNECTION.



ABOUT THE AUTHOR
Nelson Pass, the proprietor of Pass Laboratories and a frequent collaborator to *audioXpress,* enjoys creating trouble.

speaker drivers, and considerable differences of opinion exist as to the best network for any given set of drivers.

CAUTIONARY COMMENTS

Everything you read here should be taken as approximate examples to help you arrive at your own filter types and values. It is rare for a textbook filter example to be the best in any particular real-life case. The best-sounding crossover networks—whether active or passive—are usually the result of painstaking experimentation, measurement, and listening. Remember also that the typical loudspeaker driver only approximates a resistor as a load, and additionally has an acoustic output to be considered, which is not very flat or phase perfect.

The decisions about what frequency or slope are your own, and you can mix and match them as you please. Occasionally you must consider protecting a tweeter or midrange from too much power at lower frequencies, but after that you can play as you like.

INSIDE OUT

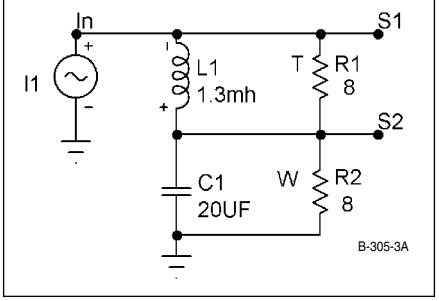
When you switch from designing filter networks for voltage sources to networks with current sources, you will find that everything is suddenly turned inside out. Capacitors become inductors and vice versa, and "T" networks become "pi" networks and vice versa. Multiple filters that used to be placed in parallel off the amplifier's output terminals are now strung in series. Don't worry, we'll get you straightened out.

Figure 2 shows part of this difference. At the top is a voltage source amplifier driving three filters and speakers in parallel to divide the audio for a woofer, midrange, and tweeter. They all share the same voltage and the same ground. At the bottom is a current source amplifier where the filters and speakers are wired in series. They all share the same amount of current.

This is a key item—that just as voltage source filters can assume a given equal input voltage, current source filters can assume equal input current. In this way, both types of filters can be designed independently so that the characteristic of one filter does not alter the characteristic of another. You can't say the same thing about voltage source filters strung in series or current source filters run in parallel—in those cases, the specifics of each filter will influence the others, as we will see in some bandpass filter examples later.

Consider some filter examples, which have been constructed for matching –3dB points at the crossover frequency, so that they present a nearly constant impedance to the source. Having a constant total impedance allows use with voltage sources equally well. Keep in mind that with a voltage source driving networks in series, when you adjust one filter, you affect the response of the others. With a current source, the response of other filters in series is not affected.

FIGURE 3A: ONE-POLE TWO-WAY FILTER.



SINGLE-POLE TWO-WAY FILTER

Figure 3A shows a simple current source crossover where the woofer and tweeter are each crossed over at 1kHz and at a slope of 6dB/octave. The values of the inductor L and capacitor C are determined by the formulas:

L = R1 / (6.28 * F) and C = 1 / (6.28 * R2 * F)

L is the inductance in henries, R is the load resistance in ohms, C is the capacitance in farads, F is the transition frequency in hertz. By the way, the "T" and "W" in *Fig. 3A* indicate tweeter and woofer. When you see an "*M*" later, it will mean midrange.

Figure 3B shows the resulting response of this, the simplest filter we will see. As will be the case of all curves, the vertical axis is in decibels (dB) and the horizontal axis is frequency.

SINGLE-POLE THREE-WAY FILTER

You can produce a one-pole three-way filter easily by simply adding an additional filter in the middle for the midrange driver (*Fig. 4A*). The formulas

FIGURE 4A: ONE-POLE THREE-WAY FILTER.

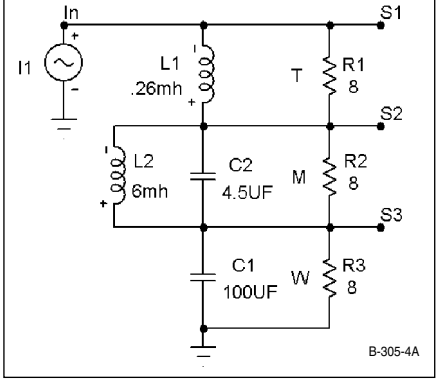


FIGURE 3B: ONE-POLE TWO-WAY FILTER RESPONSE.

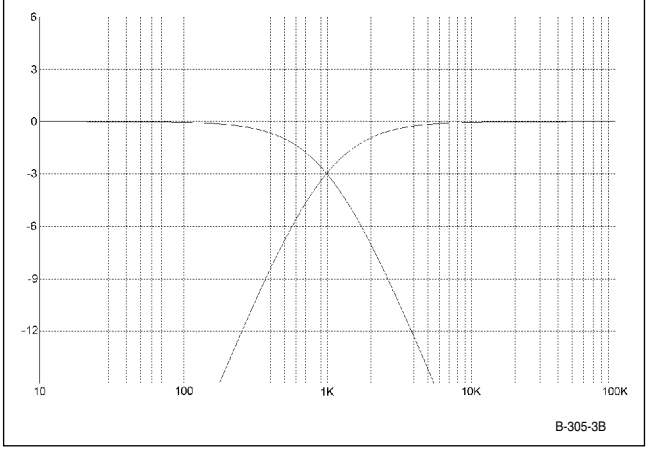


FIGURE 4B: ONE-POLE THREE-WAY RESPONSE.

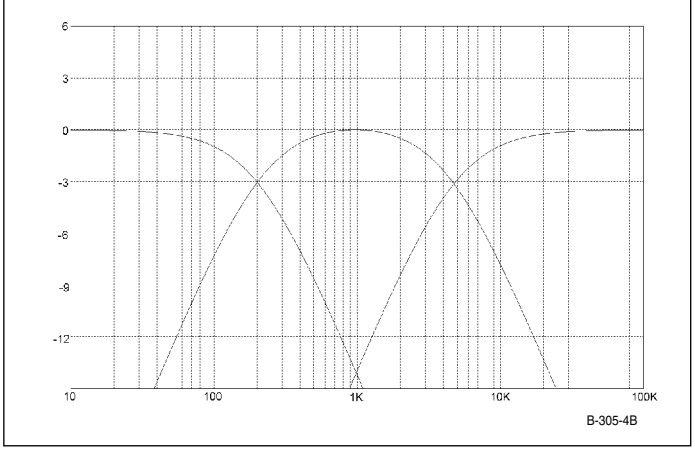


FIGURE 5A: TWO-POLE TWO-WAY FILTER.

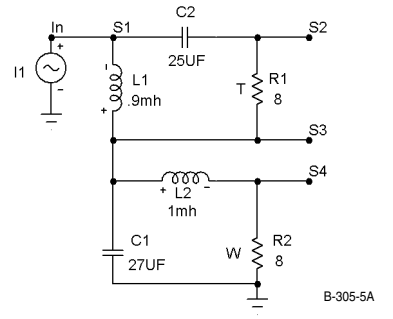
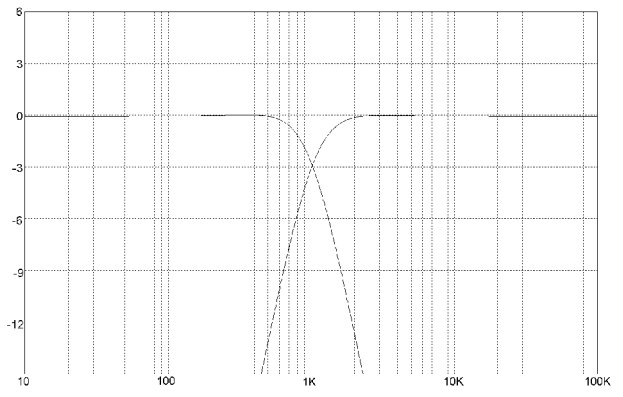


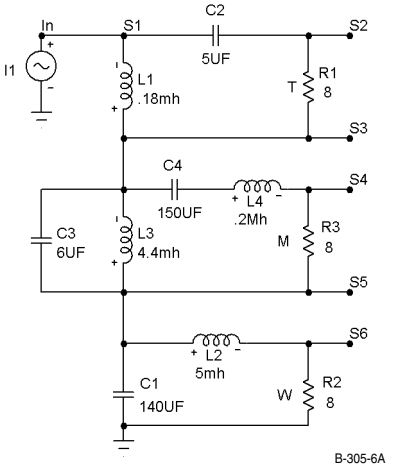
FIGURE 5B: TWO-POLE TWO-WAY RESPONSE.



applying to the previous example will work perfectly for the woofer and tweeter, but the midrange filters L2 and C2 will be off slightly, depending on the proximity of the two crossover points, which in this case are 200Hz and 5kHz. Here the coil needs to be about 10% lower than suggested by the calculation, and the capacitance needs to be about 10% higher. This is due to the loading influence of each of these elements on the other, and the variation increases as the crossover frequencies get closer.

You can calculate this if you're mathematically gifted, run a computer program if you can follow instructions, or try wiring in different values if you're good with your hands. All three approaches are likely to end up with the "wiring in different values" when it comes time to evaluate the results. In this case the results are in *Fig. 4B*.

FIGURE 6A: TWO-POLE THREE-WAY FILTER.



TWO-POLE TWO-WAY FILTER

In *Fig. 5A* we add another pole to each filter in the form of C2 on the high pass and L2 on the low pass, and we get the curves of *Fig. 5B*. The formulas have changed a bit, but you can scale from the examples. Remember that the high and low pass are independently adjustable. As you vary the frequency and resistance of the load, use the following rules:

- All examples of C are inversely proportional to the variation in frequency and load impedance.
- If you double the value of R, then you will halve the value of C.
- If you double the value of the frequency, you will halve the value of C.
- If you double the resistance and the frequency of both, you will quarter the value of C.
- All examples of L are inversely proportional to the variation in frequency, as with C, but they are proportional to the resistance.
- If you double the value of R, then you double L also.

TWO-POLE THREE-WAY FILTER

As with the one-pole three-way filter, we simply sandwich in a midrange filter in series with the high-pass and low-pass systems. The rules for the L and C values are the same as the two-pole two-way example, and of course don't work as well for the midrange depending on the proximity of the two crossover frequencies. Like I said before, you can calculate it out exactly, but you'll still end up fooling around with it when you actually adjust the speaker, so don't become too excited.

Figures 6A and *6B* show the circuit and the resulting curves. As in all the

FIGURE 7A: THREE-POLE TWO-WAY FILTER.

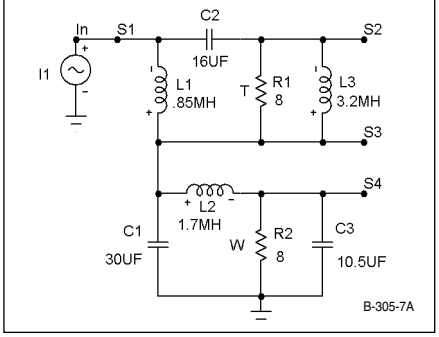


FIGURE 6B: TWO-POLE THREE-WAY RESPONSE.

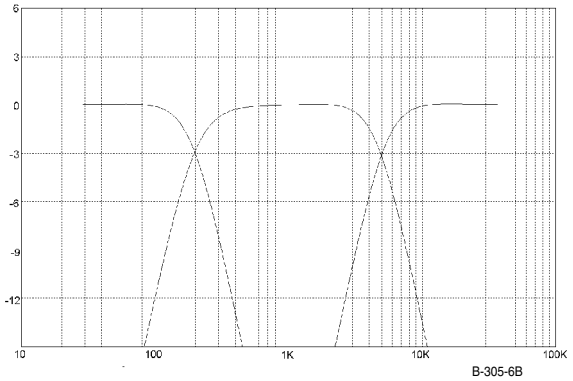
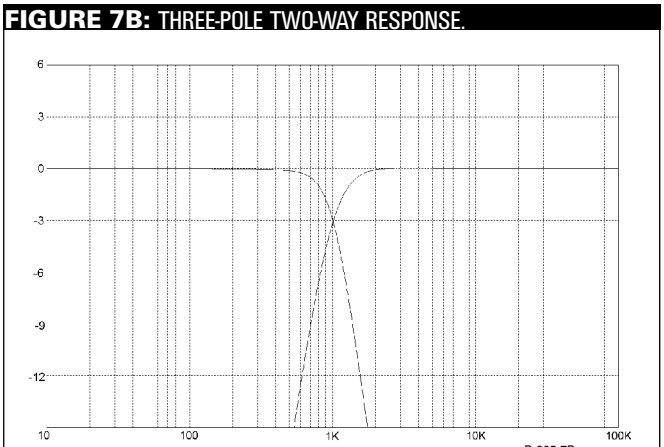


FIGURE 7B: THREE-POLE TWO-WAY RESPONSE.



other examples, the values of L and C scale with load resistance and frequency. Again, L is proportional to R and inversely proportional to F. C is inversely proportional to R and F.

THREE-POLE TWO-WAY FILTER

Now we're starting to get into some complicated stuff (*Figs. 7A* and *7B*). All the rules of the preceding examples apply, such as they are. For scaling, L is proportional to R and inversely proportional to F. C is inversely proportional to R and F.

This is a good time to note that there is no rule that says you must pick the same frequencies and slopes for adjacent elements. More often than not, I use single-pole slopes for midranges, one or two poles for tweeters, and two or three poles for woofers. Also more often than not, I set the high- and low-pass portions at different frequencies. It's strictly a matter of what works best, and that is usually not exactly the textbook version.

THREE-POLE THREE-WAY FILTER

This is the most complex example we will look at. The high-pass (tweeter) portion and the low-pass (woofer) portion are L C and R scalable as previously, and as long as the midrange filter frequencies are at a ratio of 5000Hz / 200Hz (25 to 1), then you can scale it as well. If they're not at that ratio and you really want those slopes, be prepared for some work.

Figures 8A and *8B* show the result of my labors in this regard.

FIGURE 8A: THREE-POLE THREE-WAY FILTER.

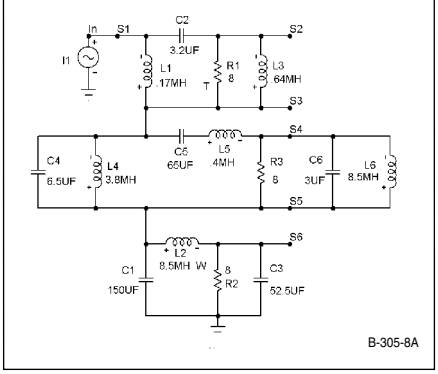
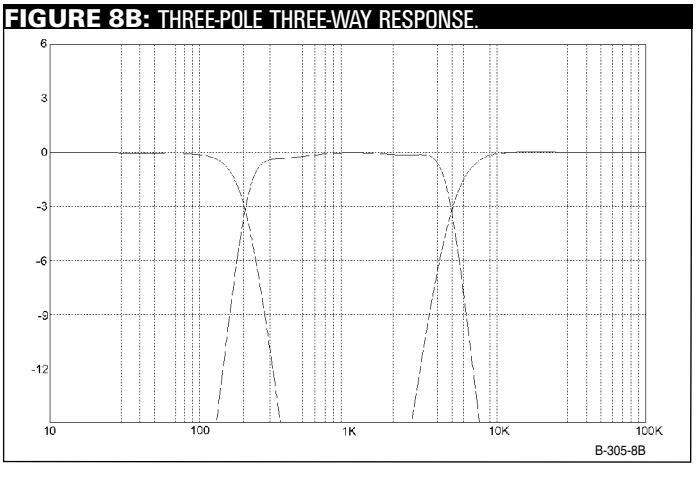


FIGURE 8B: THREE-POLE THREE-WAY RESPONSE.



CONCLUSION

The concept of filter networks in series is not new, but it is not that popular because with a voltage source, anytime you adjust one filter you affect the response of the others. With a voltage source amplifier this is avoided by placing the filter systems in parallel, so that they all share the same input voltage independently of each other.

With a current source amplifier, you can use filters in series easily because they all share the same current, and adjusting the filter values in one network does not alter the response of the other filters. You can run networks in parallel off a current source, just as you can series filters with a voltage source.

As pointed out previously, the filter examples I have given have a fairly constant aggregate impedance, so you can use them as is equally well with a voltage or current source. With a voltage source, however, the moment you adjust one of the filters, you'll find yourself readjusting the others.

Current source amplifiers aren't for everybody, although the great Malcolm Hawksford might effectively argue the case. I have certainly found that they have real charm with high-efficiency full-range loudspeakers as I have detailed elsewhere. Of course, the definition of "full-range" varies, and there are quite a few cases where users—myself included—have attached a subwoofer and/or a tweeter to extend this range. If you do, you have the choice of using active electronic crossovers or passively filtering these pieces into the system.

I have successfully driven Lowthers, Fostexes, and their cousins with current source amplifiers while applying a passive single-pole low-pass filter to them. From there I can easily bring in a tweeter (such as the TAD PT-R9 ribbon) with a passive two-pole filter, usually at 10kHz and above. Generally it's been my experience that a big subwoofer such as El Pipe-O (May '03 *aX*) prefers a voltage source for high damping, so I have used an active filter to divide it from the full-range speaker in the range below 100Hz.

The more general category of limited-bandwidth high-efficiency drivers can benefit from current source amplification as well, particularly examples of high-efficiency woofers coupled with horn-loaded compression drivers. This article was an outgrowth of that investigation, which is still progressing.

In any case, we have seen that constructing crossover filters for current source amplifiers is not particularly difficult, or at least not more difficult than for voltage amplifiers. For that matter, we shall soon see that constructing a current source amplifier is not more difficult than a voltage source.

 aX

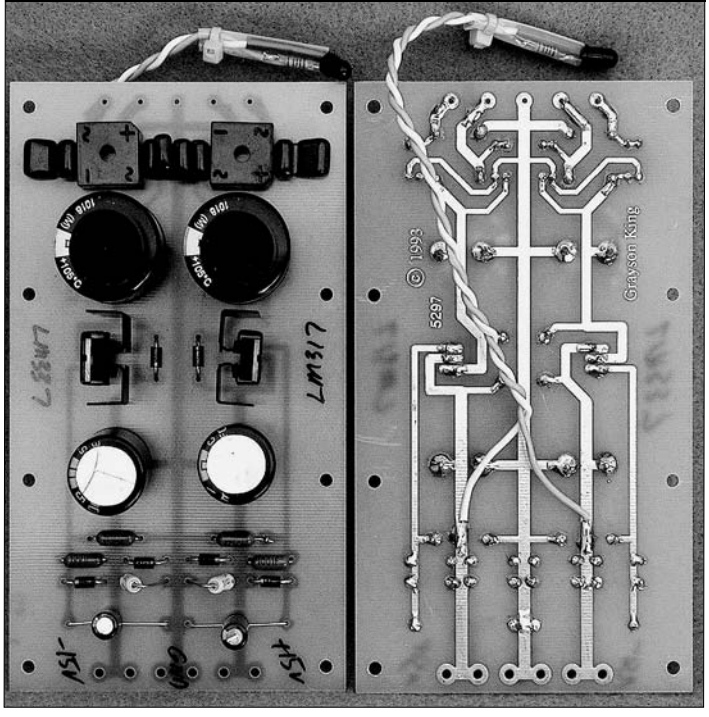
solid state

A Music Library Preamp

By Gary Galo, Regular Contributor

You can build this simple preamp control center that's used in a college music library in upstate New York.

PHOTO 1: TOP AND BOTTOM VIEWS OF THE POWER SUPPLY REGULATOR, USING THE PC BOARD DESIGNED BY GRAYSON KING. THE PILOT LED AND DROPPING RESISTOR ASSEMBLY ARE SOLDERED TO THE BOTTOM OF THE BOARD. ONLY ONE BOARD IS NEEDED FOR THE PREAMP.



Our music library in The Crane School of Music at SUNY Potsdam has ten listening rooms that receive very heavy use during the academic year. About three years ago I began to plan replacing the listening equipment in those rooms, including CD players, turntables, loudspeakers, preamps, and power amps. One problem I face whenever we replace gear used by students and faculty is finding equipment that is simple to operate. A typical Far East integrated amplifier usually has a multitude of knobs and switches on the front panel—when someone can't get it to work, it's usually because half of them are in the wrong position. Students also have a nasty habit of turning on the loudness contour and turning up the bass, assaulting the ears *and* the speakers.

I decided that the library listening rooms needed a control center that was virtually foolproof, so I decided to design and build my own purist preamp, one with only two knobs: a volume control and a selector switch. No tone controls, no loudness contour, and none of the other confusing controls often encountered on commercial equipment. Recording capability is out of the question, since library policy prohibits copying of recordings, most of which are under copyright. So, no tape outputs. System power will be controlled with a rack-mounted power strip, so there won't even be a power switch.

Simplicity is also sonically beneficial. My goal was to design a good-sounding preamp, but I made no attempt to achieve the level of performance of my recent Adcom GFP-565 remake[1,2]. That would be overkill for this application. Another benefit of simplicity is ease of construction. Builders looking for their first "from scratch" project should find this preamp a good choice. Total parts cost is around $325, including the case.

The preamp's circuitry is capable of high-end performance, limited mainly by parts quality. I did not use exotic capacitors and resistors for this preamp. Resistors are generally 1% Yageo metal film types and capacitors are Panasonic P-series polypropylenes, both readily available from Digi-Key. But, you may become as carried away as you wish with parts quality, depending on your budget and listening requirements. I used D.H. Labs' BL-1 interconnect for internal wiring on the prototype (available from Welborne Labs), but I built the ten library preamps with garden-variety wire.

If you choose to use premium internal wiring, you may also consider D.H. Labs' new Revelation Series Pure Silver Hookup Wire, or their Silver Sonic OFH-series Hookup Wire, both sold by Parts Connexion and Welborne (Welborne is selling the Pure Silver type as their Special Purchase Teflon-Insulated Pure Silver Hookup Wire). I designated this preamp the CML-1 (for Crane Music Library).

PHONO PREAMP

Our library still has thousands of LP records. Although we are continually expanding our CD collection, many needed recordings have not appeared on CD, and budgetary constraints have prevented us from purchasing as many CDs as we'd like. So, there is still a demand, albeit reduced, for LPs. Music librarian Ed Komara and I decided that four listening rooms equipped with turntables would be sufficient to satisfy the demand. So, I needed to build only four preamps with phono stages.

I designed the phono preamp using

my spreadsheet that does the Lipshitz math for all four feedback-based networks[3]. The phono preamp is a bit unusual, since it uses RIAA feedback network "C" (*Fig. 1*). Every preamp I've seen uses either network "A" or "B," and Reg Williamson also notes that he's never seen network "C" used in a commercial phono preamp[4].

Network "C" has a reputation for not lending itself to off-the-shelf component values. But, in experimenting with my spreadsheet, and checking the results with circuit simulation, I designed a network "C" that uses single-value, off-the-shelf parts, except for one set of parallel capacitors. Resistors are from the standard E96 series, and the capacitors are E12.

The RIAA network is built around Analog Devices' excellent AD745J op amp (IC1), the same ultra-low noise, FET-input device I used in the Adcom 565 mod series. The 8-pin DIP AD745JN is no longer available, but Rochester Electronics, a firm specializing in discontinued semiconductors from major manufacturers, still has 13,000 in stock. (At least one reader has noted that Rochester doesn't always respond to e-mail inquiries—call them if you have trouble). You can also use the current version, the AD745JR-16, a 16-pin SOIC package.

If you use the SOIC version, you'll also need the Aries SOIC-to-DIP adapters sold by Digi-Key. Note that the AD745KR-16 stocked by Digi-Key is the same op amp with tighter DC specifications. The op amp is buffered by a high-current IC buffer placed inside the feedback loop (IC2). R3 satisfies the gain-of-5 stability requirement of the AD745.

The phono preamp has 40dB of gain at 1kHz, which makes it suitable for a variety of medium- to high-output magnetic cartridges. I decided to deal with the DC offset using an output coupling capacitor. I used 5µF in the prototype, which puts the –3dB point at 1.25Hz, but I decided that an IEC-style rolloff would be more appropriate for the library application.

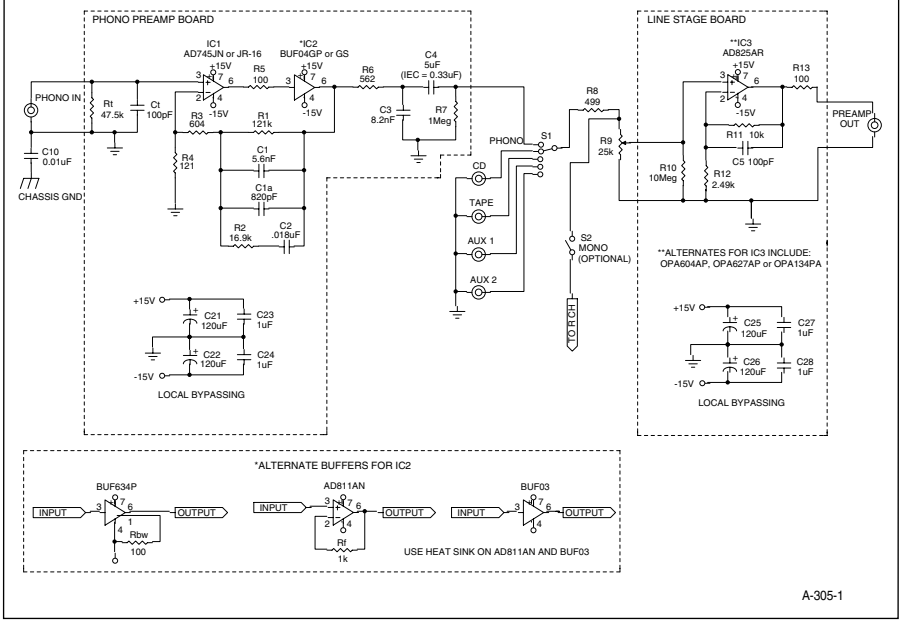
The correct value for an IEC rolloff is 0.33µF, which puts the –3dB point at 19Hz with the 25k load of the volume control. You could trim this value for the exact IEC-recommended time constant of 7950µs, or 20.02Hz, but I think the off-the-shelf capacitor value is close enough. R7 is the discharge resistor for C4, which prevents clicks and pops when you move the selector switch to the phono position.

Finding a quality 5µF capacitor in a reasonable physical size can be frustrating. Nearly all film capacitors for high-performance audio are made only in high voltages—200V to 600V is typical—since the manufacturers seem to be tailoring these products for tube designs and loudspeaker crossovers. What high-end audio builders really need is a high-quality, 50V capacitor for solid-state designs.

The 5µF capacitor I used in the prototype is a 50V polycarbonate type manufactured by Electronic Concepts, Inc., type 5MC22. This cap measures only 0.8″ long and easily fits on the boards I recommend later. Electronic Concepts also makes a 5µF/100V polypropylene capacitor, type 5MP, measuring only 1.25″ long. Contact Elcon Sales for pricing information on the Electronic Concepts capacitors.

The RIAA feedback network is low impedance, which keeps the noise levels as low as possible. My simulation circuit for measuring RIAA accuracy

FIGURE 1: COMPLETE CIRCUIT OF THE CML-1 PREAMP, ONE CHANNEL. THE PHONO PREAMP CONSISTS OF A BUFFERED AD745JN OP AMP AND RIAA NETWORK "C." THE LINE STAGE USES AN UN-BUFFERED AD825AR OP AMP.



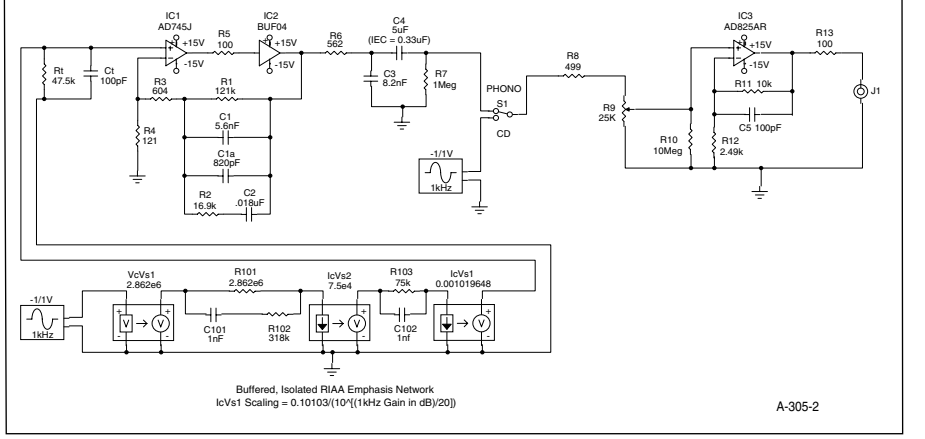
(*Fig. 2*) actually has two signal generators and a SPDT selector switch (S1), so both the line stage and phono preamp can be simulated. My schematic capture and simulation program is CircuitMaker 2000 (for more details on my simulation methods, and RIAA measurements, see Part 4 of my Adcom GFP-565 preamp series, Feb '04 *aX*).

For the RIAA simulations, the output is taken at the junction of C4 and R7, with S1 set to phono so the line stage loading is present. The simulated RIAA response, with a 5µF coupling capacitor, is ±0.014dB, 20Hz to 20kHz (*Fig. 3*). Final results will depend on resistor and capacitor tolerances.

The Panasonic P-series polypropylene caps used in the RIAA phono preamp come in 2% tolerance for values of 1000pF and higher, and 5% below 1000pF. For the prototype, I hand-selected these caps using my LCR meter (note that C3 is part of the RIAA circuit). You can upgrade the capacitors to the WIMA FKP-2 types carried by Welborne, but not without additional paralleling, since the FKP-2 caps are not manufactured in a complete E12 series, only in selected values.

You can also make the C1/C1a value of 6420pF from 4700pF + 1500pF + 220pF, and you can make the 0.018µF C2 with 10,000pF + 4700pF + 3300pF. An alternative three-cap solution to C2 is 15,000pF + 1500pF + 1500pF, but Welborne currently does not carry the 15,000pF value. Welborne stocks only 2.5% FKP-2 caps—hand selection is still recommended for the tightest RIAA accuracy.

FIGURE 2: SIMULATION CIRCUIT FOR THE CML-1 PREAMP. A MATHEMATICALLY IDEAL RIAA EMPHASIS MODEL IS USED TO CHECK RIAA ACCURACY.



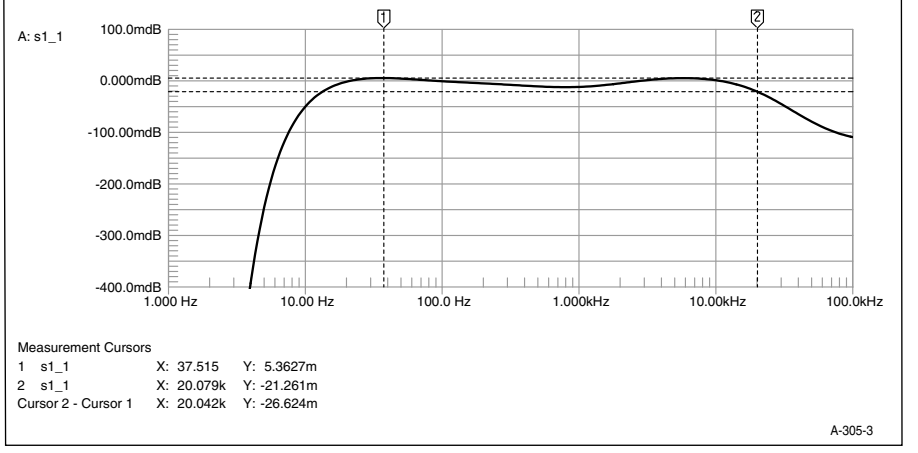
BUFFER OPTIONS

For the IC2 buffer, I originally used Analog Devices' BUF04GP, which is a high-speed, closed-loop, current-feedback device. Slew rate is 3000V/µs, output current is 40mA continuous, and noise is a very low 4nV/√Hz. The BUF04 device has now been discontinued, but there are several options. Rochester Electronics has 8,900 of the BUF04GS version in stock. The GS version is SOIC, so you'll need another pair of the Aries adapters.

There are other possibilities for IC2, including the Burr-Brown/TI BUF634P, Analog Devices' AD811AN, or Analog Devices' long-discontinued, but superb-sounding BUF03. Both the BUF634P and AD811 require a simple adaptation. You can simply drop the BUF634P into this circuit, but bandwidth and output stage biasing are at a minimum. Burr-Brown suggests adding an external resistor between pin 1 and the negative supply rail to extend the bandwidth, which also increases the idle current.

I found that 100Ω is a safe value for the BUF634P used without a heatsink; this value puts the idle current at about 10mA, close to that of the BUF04. The BUF634P is also a closed-loop device. It has a slew rate of 2000V/µs, output current of 250mA (with appropriate heatsinking), and voltage noise of 4nV/√Hz. Although Burr-Brown doesn't specifically say so, the high speed of this device indicates that it is a current-feedback design.

FIGURE 3: RIAA SIMULATION OF THE CML-1 PHONO PREAMP. RESPONSE IS ACCURATE ±0.014DB, 20HZ TO 20KHZ. THE RESPONSE WILL BE 3DB DOWN AT 19HZ IF THE IEC-STYLE ROLL-OFF IS INCORPORATED.



The AD811AN current-feedback amplifier has received much coverage in these pages. Since it's an open-loop device, the AD811AN requires a 1k feedback resistor between pins 6 and 2 for proper operation. Slew rate is 2500V/µs, output current is 100mA, and voltage noise is a very low 1.9nV/√Hz. You should always use a heatsink with the AD811AN, which idles at over 16mA on ±15V rails.

You may remember the BUF03 from my Philips DAC960 modification article in *aX*'s predecessor, *Audio Amateur*[5]. The BUF03 was designed by PMI, a com-

PHOTO 2: TOP AND BOTTOM VIEWS OF THE LINE STAGE, BUILT WITH THE OLD COLONY DG13R BOARDS AND BURR-BROWN OPA134PA OP AMPS. THE ANALOG DEVICES AD825AR OFFERS EVEN BETTER SONIC PERFORMANCE, BUT REQUIRES A SOIC-TO-DIP ADAPTER. ONLY ONE BOARD IS REQUIRED FOR THE TWO CHANNELS.

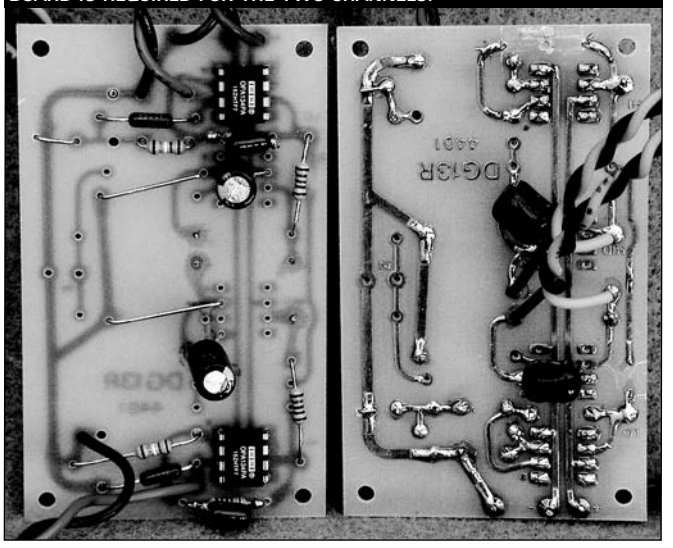
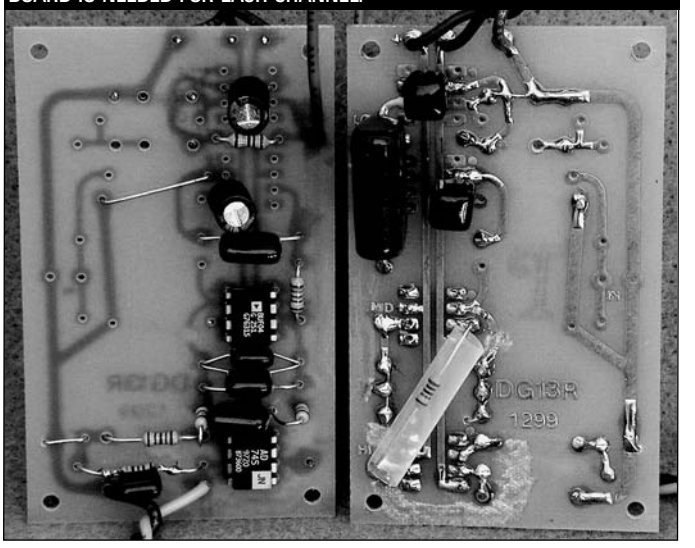


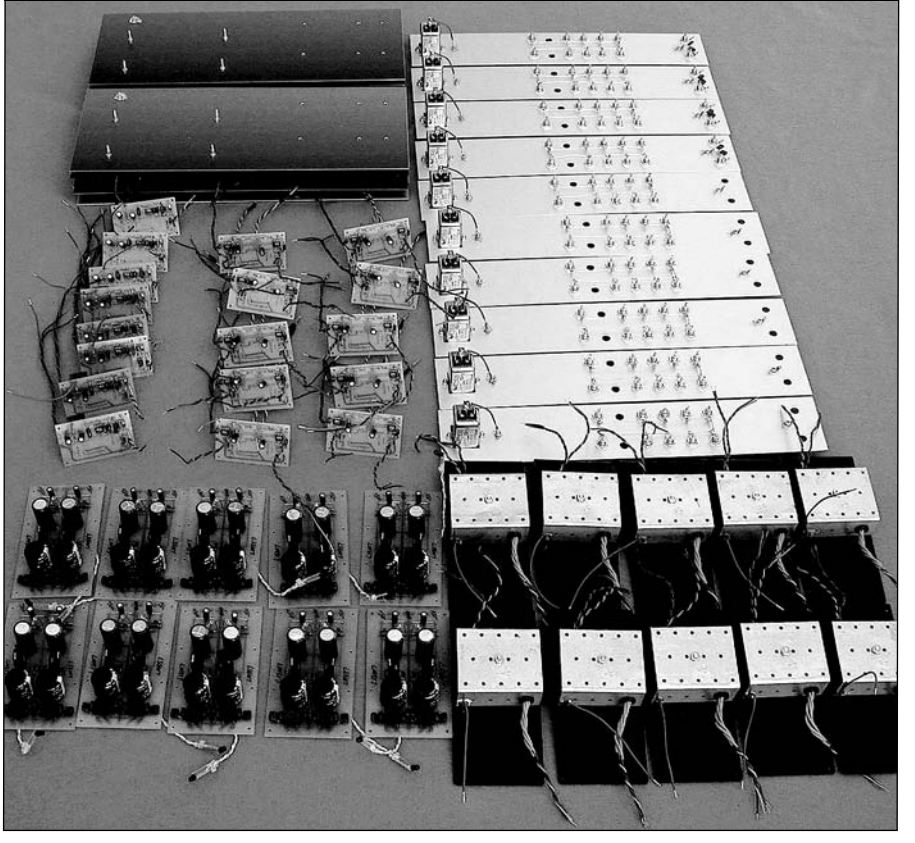
PHOTO 3: TOP AND BOTTOM VIEWS OF THE PHONO PREAMPS, USING THE OLD COLONY DG13R BOARDS. ANALOG DEVICES' AD745J FET-INPUT OP AMP IS BUFFERED WITH THEIR BUF04. OTHER SUITABLE BUFFERS INCLUDE THE AD811AN, BUF634P, OR THE DISCONTINUED BUF03EJ. ONE BOARD IS NEEDED FOR EACH CHANNEL.



pany taken over by Analog Devices in the early 1990s. This open-loop, FET-input buffer was outstanding for audio, but it was expensive to manufacture, and its TO99 case style became quite unpopular for ICs, so it was discontinued. Rochester Electronics still has over 5,700 BUF03EJ devices in stock, but they cost around $24 each. With an idle current of 19mA, this device runs hot—a heatsink such as the Digi-Key HS101-ND "gear type" is mandatory.

Slew rate is 220V/µs, voltage noise is specified at 50nV/√Hz, and the device will deliver 70mA of peak output current. The noise spec may seem high, but the BUF03EJ does not degrade the noise performance of the phono preamp when it is used inside the feedback loop with the AD745J. Since the buffer is used inside the op amp's feedback loop, it doesn't af-

PHOTO 4: ASSEMBLY-LINE CONSTRUCTION FOR THE TEN MUSIC LIBRARY PREAMPS. CLOCKWISE FROM THE LOWER LEFT: POWER SUPPLY BOARDS, PHONO PREAMPS AND LINE STAGES, BOTTOM PLATES, ASSEMBLED REAR PANELS, AND THE LEFT SIDE PANELS FITTED WITH SHIELDED, TOROIDAL POWER TRANSFORMERS.



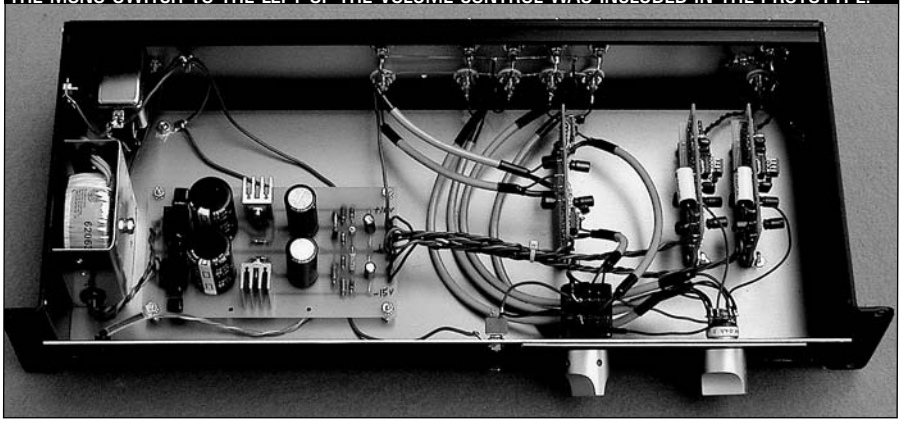
fect the RIAA response either.

Sonically, the BUF634 and BUF03 are the best performers. The BUF634 is smooth, silky, and airy in the treble region, with an extremely clean and detailed sonic presentation. The BUF03 has a beautifully liquid sound with a touch of euphonic analog warmth that is quite captivating. The treble is smooth and silky without being quite as airy and extended as the BUF634, but inner detail is still excellent.

I would say that the BUF634 is probably more accurate, while the BUF03 is just a bit more euphonic. Both chips offer a powerful, dynamic presentation, especially in the bass. The BUF04 is still a fine performer, but it isn't quite as smooth or as detailed as the BUF634 and BUF03.

Surprisingly (to me) the AD811 came in last in my listening comparisons. This device sounds a bit coarse in the upper midrange and treble, lacking the delicate, almost pristine detail of the BUF634 and BUF03. The BUF634P is readily available and will be a logical first choice for most builders. But, the BUF03 is quite remarkable in its own right, and some vinyl collectors may prefer its analog euphony.

PHOTO 5: THE ASSEMBLED PROTOTYPE PREAMP. FROM LEFT TO RIGHT: THE SHIELDED POWER TRANSFORMER, REGULATOR BOARD, LINE STAGE BOARD, AND PHONO PREAMPS. THE NOBLE VOLUME CONTROL AND GRAYHILL ROTARY SELECTOR SWITCH ARE MOUNTED ON THE FRONT PANEL. THE MONO SWITCH TO THE LEFT OF THE VOLUME CONTROL WAS INCLUDED IN THE PROTOTYPE.



LINE STAGE

I chose to keep the line stage as simple as possible, so I decided to use an un-buffered op amp (IC3) with reasonably high output current. I compared four different op amps in the prototype. Three are made by Burr-Brown/TI—the OPA134PA, OPA604AP, and OPA627AP. I also tried the Analog Devices AD825AR.

All of these chips are low-offset, low-input bias current, FET-input designs, allowing DC coupling (the OPA604 is an all-FET design). They are all capable of sufficient output current for stand-alone topologies (35mA for the OPA134 and OPA604, 45mA for the OPA627, and 50mA for the AD825).

Sonically, the AD825AR is smoother, warmer, more detailed, and more articulate than the other chips, and otherwise just simply more musical. But, it is available only in an 8-pin SOIC pack-

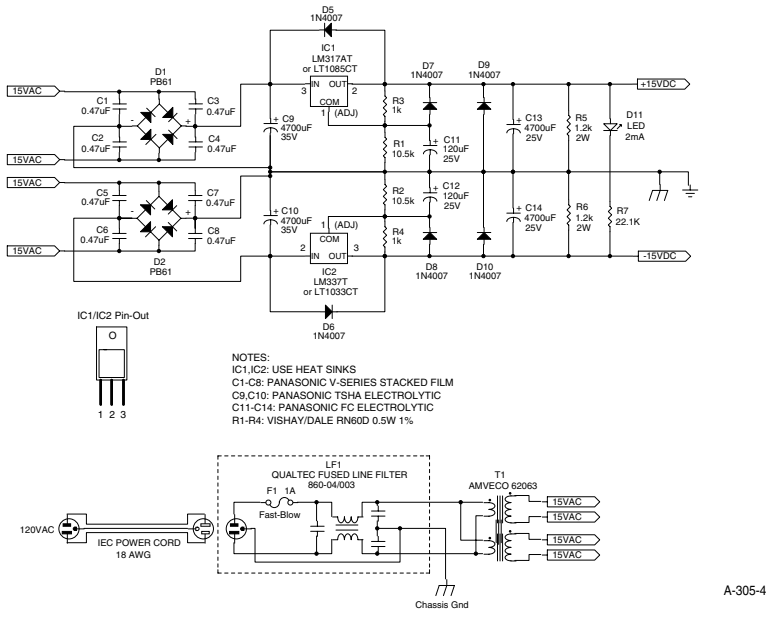
age, so you must use the Aries adapter previously mentioned. Despite this inconvenience, it is an easy first choice to get the best sound out of this preamp. I used the OPA134 in the library preamps, due to simplicity of installation and reasonable cost.

The line stage is set for a voltage gain of 5, and the 100pF feedback capacitor (C5) sets the –3dB point at 155kHz. The impedance of the feedback network is matched to that of the volume control in a normal listening position. Burr-Brown emphasizes the importance of this in the OPA134 data sheet, but correct matching is important for any FET-input op amp. Use the following formula:

$$R_S = R_1 \parallel R_2$$

where R_S is the source impedance, and R_1 and R_2 are the feedback resistors. ($R_1 \parallel R_2 = R_1$ in parallel with R_2.)

FIGURE 4: POWER SUPPLY FOR THE CML-1. YOU CAN IMPROVE PERFORMANCE BY SUBSTITUTING LINEAR TECHNOLOGY'S LT1085/LT1033 REGULATORS FOR THE GENERIC LM317/LM337 PAIR.



SOURCES

Analog Devices
(factory-direct online ordering)
www.analog.com

D.H. Labs, Inc.
612 N. Orange Ave.
Suite A-2
Jupiter, FL 33458
561-745-6406
www.silversonic.com

Digi-Key Corp.
701 Brooks Ave. South
Thief River Falls, MN 56701-0677
1-800-344-4539
www.digi-key.com

Electronic Concepts, Inc.
PO Box 1278
Eatontown, NJ 07724
732-542-7880, 732-542-0524 (Fax)
sales@eci-capacitors.com
www.eci-capacitors.com

Elcon Sales
470 Clifton, NJ 07011
973-546-5022, 973-546-5523 (Fax)
sales@elconsales.com

MCM Electronics
650 Congress Park Drive
Centerville, OH 45459
800-543-4330, 800-765-6960 (Fax)
www.mcmelectronics.com

Mouser Electronics
1000 N. Main St.
Mansfield, TX 76063-1511
1-800-346-6873
www.mouser.com

Old Colony Sound Lab
PO Box 876
Peterborough, NH 03458-0876
888-924-9465,603-924-9464, 603-924-9467 (Fax)
custserv@audioXpress.com
www.audioXpress.com

Parts Connexion
2885 Sherwood Heights Drive
Unit #72
Oakville, Ontario
CANADA L6J 7H1
order@partsconneXion.com
www.partsconnexion.com

Radio Shack
www.radioshack.com

Rochester Electronics, Inc.
10 Malcolm Hoyt Drive
Newburyport, MA 01950
978-462-9332, 978-462-9512 (Fax)
sales@rocelec.com
www.rocelec.com

Sescom, Inc.
707 N. Lindenwood
Olathe, KS 66062
1-800-634-3457
www.sescom.com

Welborne Labs
PO Box 260198
971 E. Garden Drive
Littleton, CO 80126
303-470-6585
www.welbornelabs.com

PHOTO 6: FRONT VIEW OF THE ASSEMBLED MUSIC LIBRARY PREAMP IN THE SESCOM 2RU7 RACK CHASSIS. THE PLASTIC FRONT PANEL LABEL WAS MADE BY A LOCAL HOBBY SHOP FROM A CAD DRAWING I SUPPLIED.

PHOTO 7: REAR PANEL OF THE ASSEMBLED PREAMP. I INSTALLED ONLY FOUR LINE INPUTS, BUT YOU COULD EASILY ADD A FIFTH. THE FUSED IEC POWER LINE CONNECTOR/FILTER IS ON THE RIGHT.

I decided early on to use a decent volume control, so I settled on the Noble 25k control sold by Michael Percy Audio. This pot costs $19.95, which is rather inexpensive by high-end standards, but is the most expensive single part in this preamp (except for the case). Originally, I figured that any old dual pot would be "good enough" for the library, so I used an audio taper volume control sold by Mouser Electronics for a couple of dollars. It made the preamp sound horrendous, so I decided that this was one place where I would not cut corners.

Besides, the well-made Noble pot should offer years of trouble-free performance. The 25k audio taper Noble measures around 2k in a normal listening position, so the match between the source Z and the feedback network is very good. The Noble pot has a loudness tap on each channel, which isn't used here. If the pot shaft is facing you, these are the far left terminals, which I cut off.

Input selector S1 is a Grayhill two-pole, single-deck rotary switch that should provide excellent long-term reliability. Stop pins are included, so you can set the rotation for any number of positions up to six, but I decided that five inputs would be sufficient for the CML-1. *Figure 1* shows an optional mono switch, which I included in the prototype, since I often play mono LPs. I did not put this in the ten library preamps, since the chances of it winding up in the wrong position are very high, and its usefulness would probably be limited. Since the power amp I'm using (mentioned later) has turn-on and turn-off mute, the preamp doesn't need an output muting circuit. Alternately, I recommend a sequenced power line filter.

POWER SUPPLY

The power supply regulator is based on the Walt Jung/Rich Markell design I published in *TAA* 4/90[6] (*Fig. 4*). This supply was designed around the Linear Technology low-dropout, three-terminal adjustable regulators. I used the LT1085CT and LT1033CT pair in the prototype, and recommend them for best performance. I settled on the LM317/LM337 pair for the library preamps.

The power transformer is an Amveco 62063 toroidal type—the low hum field of the toroidal transformer allowed placement of the transformer in the preamp chassis. I also put a steel shield around the transformer. With a shielded transformer, there is no trace of line hum in the phono preamp.

The adjust resistors R1 and R2 are 10.5k for the LM317/LM337 pair. If you choose the LT1085CT/LT1033CT regulators, you can add a 100Ω trim resistor in series with R1 to precisely match the positive and negative rails. This compensates for the slightly different adjust pin currents (I_{ADJ}) for the two devices. This trim is not essential, and sample-to-sample variations of I_{ADJ} in three-terminal regulators may require a different trim value if you insist on an exact match. Note that R1 through R4 must be ½W. The Qualtec AC line filter incorporates the 1A, 5mm × 20mm line fuse and a standard IEC-type, grounded power connector.

CONSTRUCTION

I used the regulator PC board designed by Grayson King for the 4/90 supply, which he used in his Valkyrie preamp[7] (*Photo 1*). The pilot LED D11 and dropping resistor R5 are attached to leads soldered to the bottom of the regulator board. Digi-Key's HS111-ND heatsinks are fine for the regulators, in most cases. Even though these are called "clip-on coolers," I recommend using thermal compound and 4–40 hardware for the best heat transfer. If you use the BUF03 or AD811 buffers in the phono preamp, you'll need larger regulator heatsinks due to the substantially

REFERENCES

1. Galo, Gary, "Adcom's 565 Preamplifier," Parts 1 through 4, *audioXpress*, Nov. '03, Dec. '03, Jan. '04, and Feb. '04.
2. Galo, Gary, "GFP-565 Preamp Follow-Up Mod," *audioXpress*, Dec. '04.
3. Lipshitz, Stanley P., "On RIAA Equalization Networks," *Journal of the Audio Engineering Society*, V. 27, No. 6, June 1979, pp. 458–481.
4. Williamson, Reg, "Understanding the RIAA Curve," *Audio Amateur*, 2/90.
5. Galo, Gary, "Pooge 5: Rite of Passage for the DAC960," *Audio Amateur*, 2/92 and 3/92.
6. Galo, Gary, "Preamp Power Supply," *Audio Amateur*, 4/90.
7. King, Grayson, "Valkyrie: A Line-Stage Preamplifier," *Audio Amateur*, 1/94.
8. Murdie, Lorelei, "The Galo-Modified GFP-565 Preamps: A Sonic Evaluation," *audioXpress*, Dec. '04.

PARTS LIST

Chassis Parts and Hardware:

(1) Rack chassis, Sescom 2RU7
(6) Gold-plated RCA jacks, black band, MCM Electronics 50–2110
(6) Gold-plated RCA jacks, red band, MCM Electronics 50–2105
(2) Control knobs, Mouser 450–3000
(1) LED mounting clip (for D11), Digi-Key 67–1044-ND

Misc.

Power transformer shield, 5″ × 3 ⅛″ steel mending plate, USP connectors #NP35, local hardware store
#8 hardware for transformer mounting, local hardware store
#8 flanged nylon bushing or shoulder washer for transformer mounting bolt, local hardware store
6–32 × ⅝″ machine screw, lock washer, hex nut, and knurled head thumb nut for ground post, local hardware store
Angle brackets with 4–40 tap (line and phono PCB mounting), Mouser 534–616
Nylon #4, 0.1150″ flat washers (line and phono PCB mounting), Mouser 561–D462
#4 × ¼″ standoffs (power supply PCB mounting), Mouser 534–398
Ground lugs, 4–40 and 6–32 machine screws, lock washers, nuts,
D.H. Labs BL-1 Interconnect and Hookup Wire (Welborne Labs; Parts Connexion)

Power Supply

(1) Valkyrie preamp power supply board, Old Colony PCBK-13B—or—Welborne PS1B power supply board
(1) Amveco 62063 toroidal transformer, 15 + 15V AC, 25VA, Digi-Key TE62063-ND
(1) Qualtec 860–04/003 fused line filter (LF1), Mouser 562-860-04/3
(1) 18AWG 3-cond. IEC power cord, Digi-Key Q102-ND—or—Radio Shack 61–2859
(1) 1A fast-acting fuse, 5 × 20mm (F1), Digi-Key F945-ND
(1) LM317AT regulator (IC1), Digi-Key LM317AT-ND—or—Linear Technology LT1085, Digi-Key LT1085CT-ND
(1) LM337T regulator (IC2), Digi-Key LM337TNS-ND—or—Linear Technology LT1033, Digi-Key LT1033CT-ND
(2) heatsinks, Digi-Key HS111-ND—or—Digi-Key HS303-ND (see text)
(2) 10.5k, ½W Vishay-Dale CMF RN60 (R1-R2), Mouser 71-RN60D-F-10.5k
(2) 1k, ½W Vishay-Dale CMF Type RN60 (R3-R4), Mouser 71-RN60D-F-1.0k
(2) 1.2k, 2W Panasonic Metal Oxide (R5–R6), Digi-Key P1.2KW-2BK-ND
(1) 22.1K, ¼W Yageo (R7), Digi-Key 22.1KXBK-ND
(8) 0.47µF/100V Panasonic V-Series (C1–C8), Digi-Key P4733-ND
(2) 4700µF/35V Panasonic TSHA (C9–C10), Digi-Key P6600-ND
(2) 120µF/25V Panasonic FC (C11–C12), Digi-Key P10270-ND
(2) 4700µF/25V Panasonic FC (C13–C14), Digi-Key P10289-ND
(2) PB61 rectifier bridge (D1-D2), Digi-Key PB61DI-ND
(6) 1N4007 diode (D5-D10), Digi-Key 1N4007DICT-ND
(1) LED, 2mA (D11), Digi-Key HLMP4700-ND

Line Stage

(1) DG13R circuit board, Old Colony PCBC-4—or—Radio Shack 276–149 or 276–250—or—MCM 21–4600 or 21–4590
(2) Analog Devices AD825AR 8-pin SOIC op amp (IC3), Digi-Key AD825AR-ND, Analog.com
(2) Aries SOIC-to-DIP adapter, Digi-Key A724-ND
(1) Grayhill 71BD30–01-Z-AJN rotary switch, 12-position (S1), Digi-Key GH7102-ND
(1) SPST toggle switch (S2), Radio Shack 275–634 (mono switch, optional)
(1) 25k Noble dual audio volume control, Michael Percy Audio
(2) 499Ω ¼W Yageo 1% metal film (R8), Digi-Key 499XBK-ND
(2) 10Meg ¼W Vishay-Dale CMF Type RN55 (R10), Mouser 71-RN55D-F-10M
(2) 10k ¼W Yageo 1% metal film (R11), Digi-Key 10.0KXBK-ND
(2) 2.49k ¼W Yageo 1% metal film (R12), Digi-Key 2.49KXBK-ND
(2) 100Ω ¼W Yageo 1% Metal Film (R13), Digi-Key 100XBK-ND
(2) 100pF/50V Panasonic P-Series (C5), Digi-Key P3101-ND
(2) 120µF/25V Panasonic FC (C25-C26), Digi-Key P10270-ND
(2) 1µF/50V Panasonic V-Series (C27-C28), Digi-Key P4675-ND

Phono Preamp

(2) DG13R circuit boards, Old Colony PCBC-4—or—Radio Shack 276–149 or 276–250—or—MCM 21–4600 or 21–4590
(2) Analog Devices AD745JN 8-pin DIP op amp (IC1), Rochester Electronics—or—AD745JR-16 SOIC op amp, Analog.com—or—AD745KR-16 SOIC op amp, Digi-Key AD745KR-16-ND
(2) IC buffers, Analog Devices BUF04GS (IC2), Rochester Electronics—or—Burr-Brown/TI BUF634P, Digi-Key BUF634P-ND—or—Analog Devices BUF03EJ, Rochester Electronics—or—Analog Devices AD811AN, Digi-Key AD811AN-ND, Analog.com
(2) Aries SOIC-to-DIP adapter, Digi-Key A724-ND (for AD745JR-16 and BUF04GS)
(2) gear-type heatsink for BUF03EJ only, Digi-Key HS101-ND
(2) 47.5k ¼W Yageo 1% metal film (Rt), Digi-Key 47.5KXBK-ND
(2) 121k ¼W Yageo 1% metal film (R1), Digi-Key 121KXBK-ND
(2) 16.9k ¼W Yageo 1% metal film (R2), Digi-Key 16.9KXBK-ND
(2) 604Ω ¼W Yageo 1% metal film (R3), Digi-Key 604XBK-ND
(2) 121Ω ¼W Yageo 1% metal film (R4), Digi-Key 121XBK-ND
(2) 100Ω ¼W Yageo 1% metal film (R5), Digi-Key 100XBK-ND
(2) 562Ω ¼W Yageo 1% metal film (R6), Digi-Key 562XBK-ND
(2) 1Meg ¼W Yageo 1% metal film (R7), Digi-Key 1.00MXBK-ND
(2) 100Ω ¼W Yageo 1% metal film (Rbw, for BUF634P), Digi-Key 100XBK-ND
(2) 1k ¼W Yageo 1% metal film (Rf, for AD811AN), Digi-Key 1.00KXBK-ND
(2) 100pF/50V Panasonic P-Series (Ct), Digi-Key P3101-ND (adjust to cartridge requirements)
(2) 5.6nF/50V Panasonic P-Series (C1), Digi-Key P3562-ND
(2) 820pF/50V Panasonic P-Series (C1a), Digi-Key P3821-ND
(2) 0.018µF/50V Panasonic P-Series (C2), Digi-Key P3183-ND
(2) 8.2nF/50V Panasonic P-Series (C3), Digi-Key P3822-ND
(2) 5µF/50V Polycarbonate (C4), Electronic Concepts, Inc. Type 5MC22 (Elcon Sales)—or—0.33µF/50V Panasonic P-Series for IEC rolloff, Digi-Key P3334-ND
(2) 0.01µF/50V Panasonic P-Series (C10), Digi-Key P3103-ND
(2) 120µF/25V Panasonic FC (C21-C22), Digi-Key P10270-ND
(2) 1µF/50V Panasonic V-Series (C23-C24), Digi-Key P4675-ND

higher current drain of these devices. I suggest Digi-Key's HS303-ND.

I mounted the power supply PC board horizontally using #4 × ¼″ standoffs available from Mouser. Welborne Labs also sells a PC board that will work with this supply—the PS1B—but fitting the larger HS303-ND heatsinks on this board may require mounting adjacent components on the foil side of the board (Welborne also offers a complete kit for their version of this supply).

You can make the line stage and phono circuits, which are relatively simple, on a variety of op-amp or prototyping boards. I used Ed Dell's very adaptable DG13R electronic crossover board for these circuits (*Photos 2* and *3*). By cutting a few traces and drilling some extra holes, you can adapt these boards for a wide variety of op-amp-based projects.

Both line stage channels will fit on one board. The phono preamps require one board per channel. Your local electronics store carries two small prototyping boards that should also work well with these circuits, 276–149 and 276–150. The latter has two traces running lengthwise down the center, which you can use for power supply connections.

MCM carries the same boards as part numbers 21–4600 and 21–4590. Again, one board should accommodate both the left and right line stage circuits, but the

TABLE1

CML-1 Phono Preamplifier Measurements

LEFT PHONO THD (W/JUNG-LIPSHITZ PASSIVE INVERSE RIAA NETWORK; 2V OUT)

	WIDEBAND	W/80KHZ LP FILTER
20Hz	0.0025%	0.0019%
1kHz	0.0024%	0.0018%
10kHz	0.0029%	0.0024%
20kHz	0.0039%	0.0034%

RIGHT PHONO THD (W/JUNG-LIPSHITZ PASSIVE INVERSE RIAA NETWORK; 2V OUT)

	WIDEBAND	W/80KHZ LP FILTER
20Hz	0.0023%	0.0016%
1kHz	0.0024%	0.0018%
10kHz	0.0027%	0.0022%
20kHz	0.0035%	0.0029%

PHONO IMD 60HZ+7KHZ (W/JUNG-LIPSHITZ PASSIVE INVERSE RIAA NETWORK; 2V OUT)

	SMPTE (4:1)	1:1
LEFT	0.0018%	0.0018%
RIGHT	0.0018%	0.0018%

PHONO SIGNAL-TO-NOISE RATIO (RELATIVE TO 2V OUT @ 1KHZ)

–94.5dB *Unweighted* (left and right channels identical)
All measurements made with Sound Technology 1700B by Gary Galo, 6/20/2004.

TABLE 2

CML-1 Line Stage Measurements

LINE STAGE THD - LEFT (2V IN > 2V OUT)	WIDEBAND	W/80KHZ LP FILTER
20Hz	0.0033%	0.0024%
1kHz	0.0027%	0.0018%
10kHz	0.0029%	0.0022%
20kHz	0.0036%	0.0027%
LINE STAGE THD - RIGHT (2V IN > 2V OUT)	**WIDEBAND**	**W/80KHZ LP FILTER**
20Hz	0.0033%	0.0024%
1kHz	0.0028%	0.0019%
10kHz	0.0029%	0.0022%
20kHz	0.0036%	0.0027%
LINE STAGE IMD 60HZ+7KHZ (2V IN > 2V OUT)	**SMPTE (4:1)**	**1:1**
LEFT	0.0024%	0.0022%
RIGHT	0.0023%	0.0021%

LINE STAGE FREQUENCY RESPONSE

–0.1dB @ 10Hz, –1.6dB @ 100kHz (left and right channels identical)
All measurements made with Sound Technology 1700B by Gary Galo, 6/20/2004.

phono preamps will require one board per channel. I mounted the phono and line stage PC boards vertically using angle brackets (available from Mouser) with one tapped 4–40 hole and one untapped hole. Put flat nylon washers on both sides of the PC boards to prevent electrical contact between the mounting hardware and the components and PC traces.

I have a few suggestions for soldering the AD745JR-16 to the Aries adapter. Cut off the four unused pins on *each* end of the AD745JR-16 op amps. These are pins 1, 2, 7, 8, 9, 10, 15, and 16. The eight remaining pins are 3, 4, 5, 6, 11, 12, 13, and 14 (these are the middle four pins on each side of the op amp). Solder these pins to the Aries SOIC-to-DIP adapter.

For some strange reason, Analog Devices made pin 3 the inverting input on the AD745JR-16. Logically, they should have made this input pin 4, so the eight pins in the middle of the package would exactly match the functions of their counterparts in the 8-pin DIP package. Pin 4 is unused, so this quirk is easy to fix—simply make a solder bridge between pins 3 and 4 of the AD745JR-16 op amp on the Aries header. Now, pin 2 on the PC board footprint will connect to the inverting input of the op amp.

CHASSIS AND GROUNDING

Sescom's 2RU7 modular rack chassis is just the right size for this project. *Photo 4* shows chassis sub-assemblies and assembled PC boards for the ten library preamps, and the inside view of the assembled prototype is shown in *Photo 5*. Pay careful attention to layout and be sure to use a star grounding arrangement. The shields from the phono input jacks go directly to ground on the phono PC boards.

I mounted the power transformer on the left side piece, on the opposite end from the phono preamps, using #8 hardware. The power transformer shield must be steel. Short of using Mumetal, the softer the steel, the better. Our local hardware store carries a 5 × 3 ⅛″ steel mending plate, USP connectors #NP35, which is exactly the right size for the transformer shield. Drill a hole dead center for the power transformer mounting bolt. Enlarge two of the existing holes in the mending plate for the transformer leads, using rubber grommets for the transformer lead holes.

It's important that the shield makes firm electrical contact with chassis ground, but not via multiple paths. I recommend running a ground wire from the shield to the star chassis ground to ensure a good connection. If you opt for this approach, you *must* insulate the transformer mounting bolt from the side panel with a flanged nylon bushing or shoulder washer, so you don't produce another ground path for the shield through the bolt and the side panel. The mending plate is galvanized—you can solder a ground wire to the plate if you scrape through the galvanized coating.

Bend the plate to form a "U"-shaped channel, and use the power transformer mounting bolt to secure the shield, using two extra nuts and a lock washer. Failure to prevent multiple ground paths from the shield may result in hum in the phono preamp, or hum on all inputs if you mount the preamp in an equipment rack. The preamp is dead silent with proper grounding.

The anodizing and paint on the Sescom chassis can prevent a good ground connection. Be sure that at the rear panel, bottom plate, front panel, and side panels are tied together electrically using ground lugs. You must scrape through the anodizing or paint in order to ensure that the ground lugs make electrical contact with the various pieces.

The wiring from the rear panel line inputs to the selector switch should be shielded. If you have only four line inputs, you can use a four-conductor shielded wire for each channel. The shield should be connected to the input jack ground bus, and can float on the other end.

MEASUREMENTS

I made all measurements on the prototype preamplifier. *Tables 1* and *2* show measurements for the phono preamp and line stage, respectively. The phono preamp measurements were made at the junction of output coupling capacitor C4 and R7, with the selector switch in the phono position, enabling the line stage loading. Distortion measurements for the phono preamp were made using the Jung-Lipshitz passive inverse RIAA network.

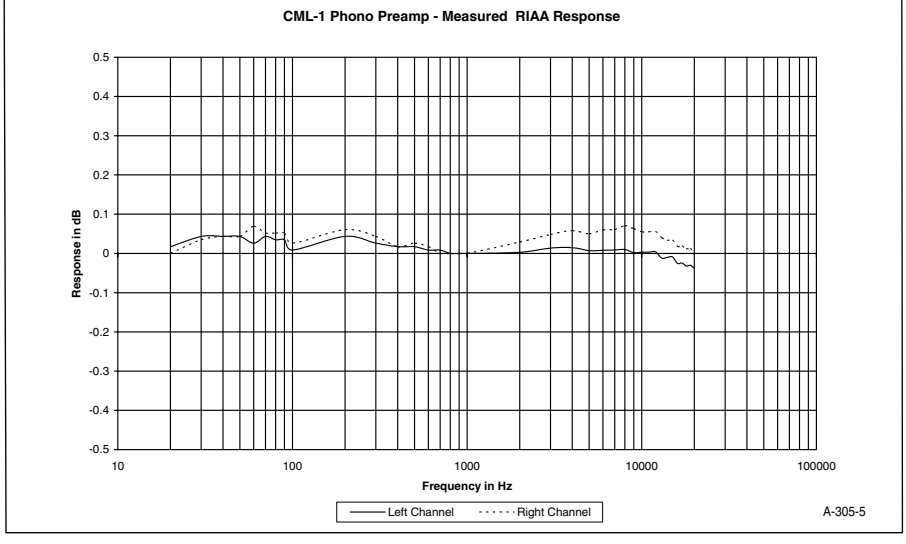
For both the phono preamp and line stage, 1kHz THD products consisted entirely of noise; 10kHz and 20kHz THD consisted of noise plus a minute amount of 2nd harmonic. Most of the distortion measurements approach the limits of my Sound Technology 1700B analyzer. The phono preamp has excellent noise performance, with noise 94.5dB below 2V out.

Figure 5 shows the measured RIAA response. I made these measurements with my modified, actively buffered Jung-Lipshitz Inverse RIAA network, using the procedure I outlined in the sidebar accompanying Part 4 of my GFP-565 series. The prototype preamp has hand-selected RIAA capacitors and C4 is 5µF. The measurements show excellent RIAA accuracy, ±0.041dB in the left channel and ±0.037dB in the right, 20Hz to 20kHz.

REMAINING COMPONENTS

After building the prototype and the ten "production" units, I completed selection of the remainder of the library systems.

FIGURE 5: MEASURED RIAA RESPONSE OF THE CML-1 PHONO PREAMP. THE PREAMP SHOWS EXCELLENT RIAA ACCURACY, WITH A DEVIATION OF ±0.041DB IN THE LEFT CHANNEL AND ±0.037DB IN THE RIGHT, 20HZ TO 20KHZ.



Cost was a deciding factor (it always is in the State University of New York), and I needed to keep the cost of one complete system, including the turntable, to under $1300. A good-sounding pair of loudspeakers could have driven up the cost considerably, but thanks to Parts Express I was able to build a remarkable loudspeaker kit for under $140 per pair. The kit is the Dayton BR-1, a vented design built around a 6½″ treated paper cone woofer with rubber surround and a 1⅛″ silk-dome tweeter.

The crossovers are made with high-quality parts, including air-core inductors and polypropylene capacitors (one large non-polar electrolytic has a polypropylene bypass), and the assembled enclosure is made with ⅝″ MDF. The performance exceeds that of assembled systems costing several times the price—anyone looking for an affordable, high-end loudspeaker should seriously consider the BR-1.

The remainder of the system includes a Sony PS-LX350H turntable ($199 including the cartridge), a Teac CDP-1250 CD player (the only inexpensive, single-play CD player with an optional rack mount; $129 plus $20 for the RM-1440 rack mount), a Samson Technologies Servo 120 Power Amp (50W per channel at 8Ω for $260), and Samson PS-9 Power Strip ($70). Samson Technologies is a division of Sam Ash, which manufactures low-cost pro-audio gear, most of which is made in China. The Servo 120 power amp is based on a Sanyo STK4038II power amp module with DC-servo control.

The toroidal power transformer is quite massive for an amp in this power class, and accounts for most of the amp's 15.6 lb. The amp offers really decent performance for the price, is very well-made, and I expect it to be very reliable. The PS-9 power strip includes surge suppression. At this price you don't get sequenced power switching, but the Servo 120 has turn-on and turn-off muting, so sequenced powering of the system is unnecessary. Performance versus cost of the completed systems is quite remarkable, and faculty and students who have heard the prototype system have been very impressed.

THE SOUND

A while back, Lorelei Murdie listened to my prototype CML-1 preamp on her system, comparing it to her slightly modified Adcom GFP-565 preamp (her preamp is stock except for LT1122 op amps in the line stage, as noted in her write-up of my 565 preamp mods[8]). Lorelei found the CML-1 preamp to be smoother and warmer-sounding than her GFP-565, with greater inner detail, a deeper soundstage, and greater extension in both the bass and treble regions. I basically agree with her reactions, and consider the mission accomplished as far as realizing my sonic goals for this preamp.

I encourage readers to use these circuits as a basis for departure, particularly in the area of parts quality. The Vishay-Dale CMF Type RN60 resistors carried by Mouser and Welborne Labs are an obvious—and inexpensive—upgrade, as are the Wima FPK-2 polypropylene capacitors already mentioned. The phono circuits would make an excellent outboard phono preamp for an existing system, and there are many other possibilities for tailoring these circuits to complement your listening requirements and associated equipment. ***aX***

speakers

A Simple Audio System
Part 2: Speakers

Back

By Aren van Waarde

For this author's simple tube amp, you need efficient, full-range speakers, which he discusses in Part 2 of this series.

In part 1 of this article (a*X* 2/05), I described a single-ended single tube amplifier which is very simple but nevertheless produces excellent sound. An Italian manufacturer produces a similar amp called "Lo Scherzo" because of both its simplicity and its musicality. ("Scherzo" means "joke," but is also a musical term.)

As I mentioned in part 1, fleapower SE amps require efficient full-range speakers to sound their best. The absence of a crossover makes such speakers easy to drive, and the single drive unit behaves almost as a point source, which may result in excellent imaging.

For my study system, I adapted a project from the French magazine *L'Audiophile*. The Parisian designers used a single 4″ driver of Japanese origin, the Fostex FE103 Sigma (*Photo 1*), in a column-shaped vented enclosure[1].

LA PETITE

My interest in this French speaker was roused for several reasons. The different incarnations of the 4″ Fostex unit (FE103, FE103 Sigma, FE103E) have been extremely popular among Japanese audiophiles since the early 1960s. Published speaker projects involving this driver are innumerable. In the pages of *L'Audiophile*, Gérard Chrétien raved about the sonic qualities of the little Fostex unit[2]. When my compatriot Peter van Willenswaard (of Audionote fame) heard the FE103 Sigmas at a show in Paris, he was equally enthusiastic[3].

The German audio amateur Klaus Schiffer has built a pair of Audiophile speakers and uses vintage Leak amplifiers to drive them. On his website, Klaus praises the little speaker as much as Gérard and Peter[4]. Finally, Bert Doppenberg, the driving force behind BD Design—a small company producing horn speakers—published a unique DIY project that employs two of the small Fostex drivers per channel[5]. This tapered quarter-wave pipe design has been favorably received by many audiophiles, both in Europe and in Canada[6].

PHOTO 1: FOSTEX FE103 SIGMA UNIT, AS USED IN LA PETITE AUDIOPHILE

It occurred to me that a driver produced in large quantities for four decades (!) and with many admirers in three continents must be really something. Man-

FIGURE 1: IMPEDANCE CURVE AND OPEN-BAFFLE RESPONSE CURVE OF THE FE103 SIGMA (ON-AXIS, 30 AND 60° OFF-AXIS). MANUFACTURER SPECIFICATIONS.

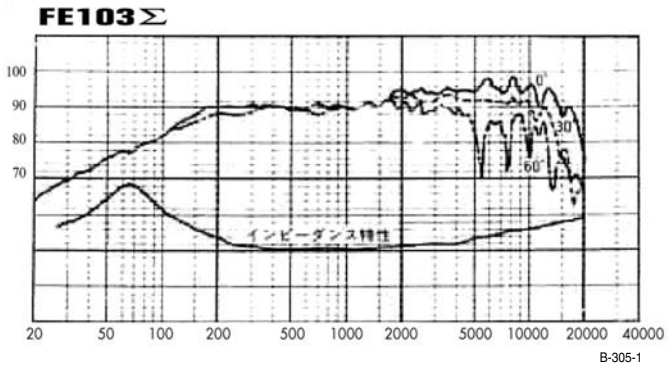


TABLE1

Data of the Fostex units

	FE103 SIGMA[1]	FE103E[2]	FE108E SIGMA[2]
Cone material	Banana fiber	Banana fiber	Banana fiber
Fs (Hz)	79	79.5	77
Re (Ohms)	7.3	7.45	6.8
Z (Ohms)	8	8	8
Qms	2.8	2.87	7.79
Qes	0.29	0.40	0.32
Qts	0.26	0.36	0.30
Vas (liters)	6.2	6.9	5.7
Mms (grams)	2.9	2.1	2.7
Sd (cm^2)	55	50.3	50
RMS power max. W	6	5	8
Music power max. W	20	15	24
Xmax (mm)		0.35	0.28
Cms (mm/N)	1.43	1.94	1.59
Frequency range	F_0 – 18kHz	F_0 – 22kHz	F_0 –23kHz
Efficiency (dB/W)	90	89	90
Magnet weight (g)	386	193	400
B_L (Tesla/m)	5.75	4.42	5.2
Efficiency N_0 (%)	0.98	0.78	0.83
Total weight (g)	960	630	1,200
Recommended	Horn or bass reflex	Horn or bass reflex	Horn loading
Price	About $40	$31.50	$85

1 As measured by the German audio magazine *Klang & Ton*
2 Manufacturer specifications (Fostex, Japan)
(Some data may have rather wide tolerances, e.g., ±15Hz for f_s)

ufacturer specifications of the FE103 Sigma are shown in *Fig. 1* and *Table 1*.

VARIOUS CABINET SHAPES

Two differently shaped vented enclosures were published for the Fostex FE103 Sigma in *L'Audiophile*. The internal volume was 26.5 ltr in both cases, with a vent of 140mm long and 70mm diameter, resulting in a cabinet resonance at 47Hz.

The initial cabinet design was very solid (the lower part of the front panel was 50mm thick!), but the shape of the enclosure was not appealing (546 × 352 × 227mm, height × width × depth[2]). When my wife saw the picture in the magazine, she called it the ugliest speaker she had ever seen! (I disagreed, for some of my own creations also do not deserve a prize for their good looks.)

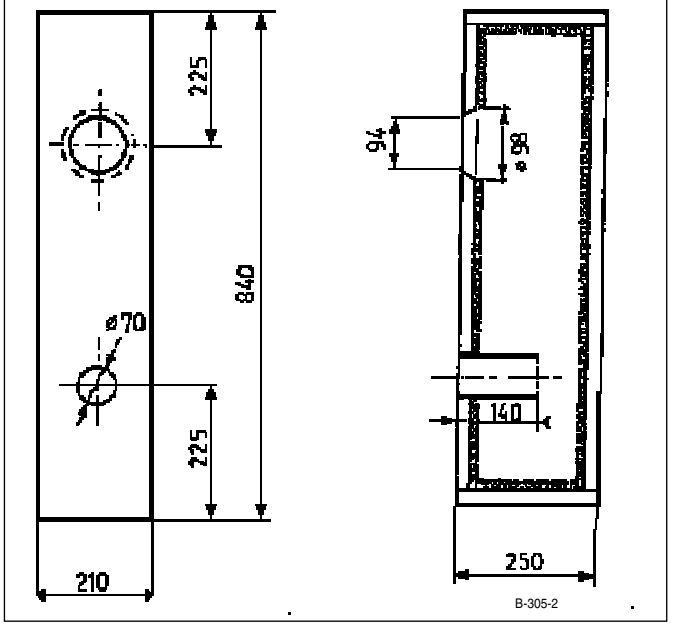
Apparently, many readers in France were of the same opinion as my dear spouse. For after three years, *L'Audiophile* published another design with identical cabinet volume and vent, but much more appealing proportions (*Fig. 2*). This design was called "La Petite Audiophile en colonne" (The Small Audiophile in column shape). I decided to build that version, not only to ensure domestic peace, but also because I concur with my wife's view.

VARIOUS MODS

The Parisian designers used medium-density fiberboard (MDF) with a thickness of 25mm (1″) to construct their enclosures. This material proved difficult for me to obtain. DIY stores in Holland only have MDF of 19mm (0.75″) in stock. They can sell the 1″ variety, but you need to special-order it in fairly large quantities, and it takes several weeks to arrive. In Germany, Klaus Schiffer encountered the same problem.

Therefore, Klaus decided to make his own pair of speakers from 0.75″ MDF and to glue a sandwich of aluminum-rubber-aluminum to the inside of each cabinet wall. To keep the internal volume at 26.5 ltr and to maintain the rigidity of the structure, I decreased the

FIGURE 2: ENCLOSURE DIMENSIONS ("LA PETITE AUDIOPHILE EN COLONNE").



outside dimensions of the enclosure a little (4mm along every axis) and glued bituminized lead sheets to the inside of each cabinet. This damping material (thickness 4mm) is sold by DIY speaker stores and is also used for panel damping in cars. The use of these self-adhesive sheets results in a very heavy enclosure which is completely "dead" in acoustic terms (i.e., free of undesired panel resonances).

The people at *L'Audiophile* used a 10mm layer of wool, glued to the inside of each cabinet wall, as stuffing material. Klaus employed the kind of wool that mechanics use for cleaning up oil. I used a dual layer of the loosely woven material (insulating cloth) that home decorators put on floors under the carpet.

For speaker wire, I used two 4m sections of old cable (Van den Hul M.C. CS 122). This is heavy cable from a Dutch manufacturer, a twinlead type with two multiwire strands of 0.16″ diameter (silver-coated copper). The Van den Hul wire may be overkill, but I thought a good cable wouldn't hurt. The wire, which I used when I was a graduate student, must be at least 15 years old.

THE SOUND

Can a 4″ driver with a power handling of less than 10W and a price of $40 (US) produce decent sound? Let other people answer this question. In the February 1984 issue of *L'Audiophile*, Gérard Chrétien wrote:

"It is almost embarrassing to talk about listening to this little speaker. The results which are obtained are so extraordinary in comparison to the small cost and the diminutive size of the driver that it is difficult to be credible in writing. You must listen for yourself.

"The overall homogeneity is excellent. The bass is there—although of course it can never have the energy of a 15″ woofer—and it lends a very good balance to the sound. However, it is not the quality of the bass which this little speaker can produce which surprises most, but rather its extraordinary capacity for resolution, and this throughout the whole audio spectrum. Whatever is the musical message, one perceives a multitude of information, of small details. The absence of filtering and the direct coupling of driver and amplifier may certainly be related to this surprising definition.

"Dynamics of the sound are very good, extremes of audio level are perfectly respected without excessive compression during heavy modulations.

"Sure, the treble is a little bit truncated; one cannot have everything! But it is of excellent quality, sweet and refined at the same time."

Besides this French verdict, I quote a German authority. Klaus Schiffer wrote in 1998:

"I have listened to this speaker for five years now. Previously, I used the KEF Calinda (also DIY). A few weeks after I changed to the Petite, I considered switching back because I was missing some bass. As soon as I heard the Calinda, I understood the strength of the Petite.

"It is the fantastic localization of each instrument or voice. The midrange and highs are superb; even the bass has a wide range. For such a small speaker, it is really good. OK, you will not feel any punch in your stomach, but believe me: you will be impressed."

I can add very little to these comments. When the Petite is driven by my own SE tube amp (see Part 1 in *aX* 2/05), a bass deficit is evident in some recordings, but this is due to my use of output transformers from old radios. When I use the (expensive) SE tube amp from my living room, or a good solid-state amplifier (Van Medevoort PA 222), the bass is more in harmony with the rest of the audio spectrum. The statements of Gérard and Klaus about dynamics, resolution, and imaging are completely to-the-point. *Photo 2* shows the homemade speaker pair.

ADDITIONAL FILTER

Some users of the Petite Audiophile considered the sound too bright, with too much emphasis on the treble end of the audio spectrum. This may be the case especially when the listening room is "hard" in acoustic terms (large surfaces of glass in several walls, stone floor, concrete ceiling, little furniture). In my study, I did not encounter this problem, because two walls are covered with

bookshelves and there are many chairs and writing desks in the room.

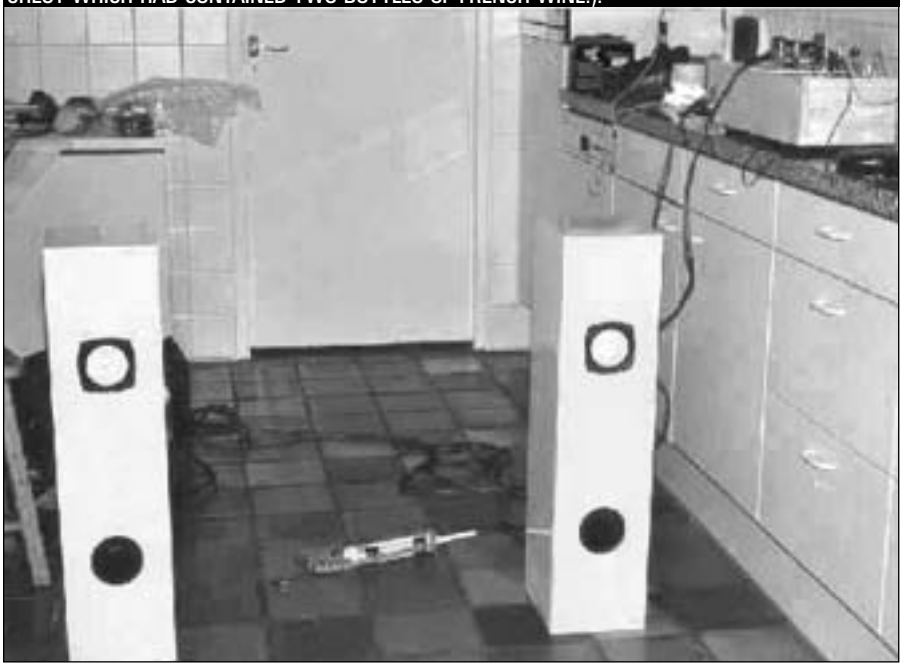
If you are less lucky, you may consider wiring a parallel connection of a 0.68mH inductor and a 4.7Ω 10W resistor in series with the speaker. Use the best quality of components you can get. The filter does not affect the low-frequency part of the audio spectrum (the region below 300Hz), but it will attenuate the treble region above 3kHz by about 4dB. You may, of course, also use your amplifier's tone controls, if present, to reach the same goal.

Klaus Schiffer used the treble filter for a while, but he finally removed it because the RL network seemed to result in loss of resolution.

PLACEMENT

How should you place these speakers so that they sound their best? I have tried several possible locations: in corners, close to the walls, and free-standing within the room. Moreover, I tried various vertical positions: floor-standing, on low stands (lifted about 20″ from the floor), and on high stands (lifted 32″ from the floor). Finally, I tried different angles with respect to the long axis of the room: axis-parallel, or toed in toward the listener.

PHOTO 2: FINISHED SPEAKERS DURING AN INITIAL LISTENING TEST IN OUR KITCHEN, WITH PROTOTYPE OF THE SE TUBE AMP ON THE DRAINING BOARD (QUITE APPROPRIATELY BUILT INTO A CHEST WHICH HAD CONTAINED TWO BOTTLES OF FRENCH WINE!).



Within my study, these speakers seem to behave in the same way as mini-monitors in general. The spatial resolution is optimal when the center of the 4″ unit is at the same height as the listener's ears, and the speakers are toed in toward him (or her). Placement

close to corners or walls results in a loss of front-to-back information; therefore it's better to avoid such positions. However, it is not necessary to place them very far from the side walls of the room; a distance of 20″ seems to be sufficient.

Thus, I finally settled on the 20″ stands, a position halfway along the long axis of the study, about 20″ from each side wall, and toed in toward the listener.

To my surprise there is not a narrow "sweet spot." Although cone speakers are said to beam the sound much more than dome tweeters, I can walk to many positions throughout the room and still perceive a good soundstage. Even if I stand between, or at the back of the speakers, the position of soloists or instruments remains well-defined. For such cheap speakers, this is no mean feat.

Like most mass-produced units, the speakers do not immediately sound their best. There is a break-in period of a few weeks, during which the compliance of the cone surround increases and the resonance frequency declines. When I had just finished the speakers, I enjoyed the excellent resolution, but I thought they were too bright sounding. However, the tonal balance improved during the first month of listening. With most recordings, they now sound completely OK, as Gérard Chrétien and Klaus Schiffer predicted.

BAD AND GOOD NEWS

Unfortunately, the Fostex FE103 Sigma unit has recently been discontinued. Several European (and Japanese?) distributors still have some stock, but when these are sold, the legendary unit will become history.

There is no exact replacement for the FE103 Sigma, but Fostex now produces a unit called FE103E and a more expensive 4″ speaker called FE108E Sigma (*Table 1*). According to Wolfgang Vollstädt from Spectrum Audio (Bremen, Germany), you can use the FE103E in an enclosure designed for the FE103 Sigma—this will result in a little more bass, a little less treble, and a slightly lower efficiency.

However, I advise you to follow another course. In the US, Fostex loudspeakers are distributed by Madisound[7], whose website[8], includes plans for gorgeous horn speakers using a single FE103E or a FE108E Sigma per channel. The neat thing about these folded, backloaded horns is that you can make them fairly easily: there are no oblique woodcuts and all panels are mounted at angles of 90°. The outside dimensions and proportions of the enclosures are similar to those of the Petite Audiophile: 840 × 180 × 405mm for the FE103E and 900 × 200 × 445mm for the FE108E Sigma (height × width × depth).

Cutoff frequency of the horns is set at approximately 50Hz, resulting in a quite extended low-frequency response (for such small drivers). The narrow baffle results in excellent imaging and the horn loading in high efficiency and good dynamics.

If I had to start all over again, I would build the recommended enclosure for the FE108E Sigma. Its published frequency response and that of the "Petite" are quite similar, although the response curve of the novel unit is more even (without any appreciable peaks or dips throughout the whole audio range).

My experience with Fostex loudspeakers indicates an excellent relationship between quality and price. If you give them a try, I don't think you will be disappointed.

Part 3 (the final part of this series) will describe a "cheapskate class A amp" with only three bipolar transistors per channel. . . for the tube-haters among you. ***aX***

REFERENCES

1. "La Petite Audiophile en colonne" (rubrique On en parle) *L'Audiophile,* no. 41, p. 80-81, 3e trimestre, 1987.
2. Gérard Chrétien, "Le Fostex FE-103 Sigma," *L'Audiophile,* no. 31, February 1984. See also: "Le Fostex FE-103" (rubrique On en parle), *L'Audiophile,* no. 29, p. 95-96, September 1983.
3. Peter van Willenswaard, *Audio & Techniek,* October 1984.
4. Klaus Schiffer's website: http://members.aol.com/ IKSchiffer.
5. You can download plans from http://www.bd-design.info/tqwp.zip.
6. See the DIY section at the Single Driver website (James Melhuish), http://melhuish.org/audio.
7. Madisound Speaker Components, Inc., 8608 University Green, PO Box 44283, Madison, WI 53744-4283.
8. http://www.madisound.com.

Product Review

Building The Vifa/Dayton Aluminum MTM Kit

By Edward T. Dell, Jr.

PHOTO 1: THE ASSEMBLED MTM UNITS ARE QUITE HANDSOME, EVEN WITHOUT THEIR SUPPLIED GRILLES. THE KIT ASSEMBLES EASILY IN NO MORE THAN FIVE TO SIX HOURS.

Parts Express
725 Pleasant Valley Dr.
Springboro, OH 45066-1158
1-800-338-0531
Fax: 1-937-743-1677
www.partsexpress.com
Part no.: 302–940, black
Dimensions: (W × D × H) 9 × 13.5 × 22″
List price: $799. MSRP $493.62/pair.
Available: cherry, beech, gloss black finishes
Impedance: 4Ω
Frequency response: 40Hz–35kHz
SPL: 86.5dB
Power handling: 150W

Over the last decade Parts Express has become a major player in the loudspeaker market both at the consumer and OEM level. Their retail interest reflects a strong commitment to kit projects for the amateur builder, and this mid-tweeter-mid (MTM) D'Appolito-style bass-reflex kit (*Photo 1*) is a good example.

The kit comes completely finished with a mounted front panel and central interior sub-panel, making the space inside the relatively small 1ft^3 cabinet a bit awkward to work in. The cabinets are constructed of ¾″ high-density particleboard, beautifully finished with either cherry or beech veneers or a high-gloss piano black lacquer.

CONTENTS

Parts Express does a fine job of packing the kit, which arrives in three boxes (*Photo 2*), two for the cabinets and one for the remainder of the parts. The kit is complete (*Photo 3*), with sheets of foam for lining half the inner cabinet surfaces, four 7″ Dayton mid-bass aluminum cone drivers, two Vifa D25AG35–06 aluminum diaphragm tweeters, two 2½″ diameter adjustable port tubes, an 11-page manual, etched circuit cards for crossover assembly, bags of crossover components, driver sealing caulk, "stakon" terminated wire harnesses, speaker edge caulk, screws, and a tube of mounting adhesive. The builder needs only to add solder and possibly hot melt glue if s/he prefers that method of mounting the 2kHz crossovers.

The manual lists required simple tools and provides an assembly order that makes good sense. An inventory of included parts allows you to check for missing parts and a number to call for remedies. You'll need a well-padded workplace to preserve the very high-quality finish. The manual combines some photos from other kits which are offered in unfinished form and without the mounted central baffle and front panel. This is slightly confusing, but not fatal. It does make some of the instructions irrelevant, however.

PHOTO 2: THE KIT ARRIVES IN THREE COMPETENTLY PACKED CARTONS, TWO FOR THE CABINETS, THE OTHER FOR ALL THE PARTS.

The very handsome cabinets are drilled for four plastic grille-mount posts, which are the first installation task. The cutouts for mounting the two 7″ aluminum mid-woofers are rabbeted to seat the rims flush with the front panel, as is true of the Vifa tweeter mounting (*Photo 4*). This minimizes the possibility of diffraction problems. The straight (non-flared) telescoping port tubes must be adjusted and glued to the correct length, mounted with #6 ¾ screws, to effect a tuned box format.

The pairs of supplied binding posts are very high-quality, and you can mount them directly through the rear cabinet wall on ¾″ centers (*Photo 5*) with

a hammer, after removing the main nut used to secure the input speaker wire. Use the nuts, lock washers, and a solder tab to secure this hardware inside the cabinet wall. A hollow nut driver can help here, but gas pliers or an adjustable wrench also work, but with greater difficulty. Bend the solder tabs outward to make soldering to them easier. You will need 2″ thick padded blocks to support the cabinet for work on the front panel to avoid stress on the binding posts.

PHOTO 3: THE KIT CONTENTS WHERE EVERYTHING EXCEPT SOLDER IS SUPPLIED. THE WIRING HARNESSES COME CUT TO LENGTH AND SUPPLIED WITH "STAKON" PUSH-ON TERMINALS.

The kit contains pieces of expanded foam with an undulating surface to damp internal reflections. The manual suggests covering half the eight surfaces formed by the central extra baffle mounted mid-cabinet. Use a sharp knife or sturdy scissors to cut the four pieces to fit. I installed the foam as instructed, since in building a kit for review I always follow instructions exactly.

KIT ASSEMBLY

If I were assembling this kit independently, I would advise a variation of the instructions. According to the manual, you should mount the crossover in the bottom-rear section, just below the input terminals. This makes the board very hard to reach. I checked the wire supplied and such an alternate location in the front floor section will work fine, and would be much easier to reach in final assembly. This would mean that the foam piece on the top of the cabinet would be installed in the front section and that for the bottom would sit in the rear. I doubt the change would make any acoustical difference in performance.

PHOTO 4: THE CABINETS FOR THE KIT ARRIVED FULLY ASSEMBLED OTHER THAN INSTALLING THE SNAP-ON GRILLE MOUNTING POSTS. THE GRILLE COMES PRE-ASSEMBLED, AND SNAPS ON FIRMLY.

PHOTO 5: HIGH-QUALITY INPUT TERMINALS ARE MOUNTED ON THE LOWER BACK OF THE CABINETS BY FIRST REMOVING THE NUTS FOR SECURING THE SPEAKER CABLES AND USING A HAMMER TO PUSH THEM SNUG TO THE BACK PANEL. I USED A PLASTIC-FACED HAMMER FOR THE JOB.

PHOTO 6: ASSEMBLING THE CROSSOVER IS SIMPLE AND WELL ILLUSTRATED IN THE MANUAL. I LEFT LABELS ON THE COILS UNTIL FINAL SOLDERING TO MAKE SURE I HAD LOCATED THE COMPONENTS CORRECTLY. THE COMPONENTS APPEARED TO BE VERY HIGH QUALITY. AN ASSEMBLY VISE AIDS ASSEMBLY AND SOLDERING.

(This might cause some interaction between the crossover coils and the driver coil, but I did not test this arrangement.)

You are asked to re-shape the supplied caulking into strands about ⅛″ in diameter to be placed around the outer rims of the drivers. The photo is a bit confusing since it shows the caulk placed, not on the outer rim, but nearer the supports for the magnet/coil assembly. I placed the caulk near the edge to make an airtight seal for the drivers. You want all the internal air pressure to go through the port, of course.

Assembling the crossovers is simple and well-supported by the instruction photos. The circuit boards are easy to use and well laid out. Some of the capacitors are longer than the hole-spacing, requiring careful shaping of the leads in order to fit snugly against the board. I followed my usual practice of skinning any oxidation from component leads by carefully scraping them with a wire cutter. You must also remove the insulation coating on the coil lead wires by scraping with a sharp knife.

You first glue the three coils, four capacitors, and two resistors to their proper locations. The coils are further secured using wire ties, a great invention (*Photo 6*). The manual has several photos clearly showing where components are located.

Good, slow, careful soldering is mandatory, as always. Double-check your work to avoid later changes to boards already glued in place. Note carefully the connector terminals on the board. The inputs are clearly marked for polarity, as are those for the tweeter.

The four mid-woofer terminals could be confusing. Those nearest the corner are the two negative connectors, the two toward the center are the positive terminals. It is possible to wire incorrectly here as I discovered the hard way. In a test I naturally had sound from the tweeter only.

Because my kit came with the front panel already installed, I found it easier to mount the drivers last. In preparation for that, I carefully separated the handy wiring harnesses for the three drivers and the one for attaching the crossover to the input terminals. I identified each for its destination since the wires' lengths vary. It is imperative that you fully illuminate the cabinet interior for the final assembly (*Photo 7*). A flashlight helps as well.

I soldered the input wiring to the terminals first. Next I connected the harnesses to their respective lugs on the crossover board. Then I glued the crossover in place (*Photo 8*). This procedure is much easier for this kit than the instructions' advice to mount the internal baffle and drivers in the front panel beforehand. If the front panel was not installed, the manual's instructions would work well.

However, all went smoothly after I reconnected the woofer wiring correctly, and the full sound of the pair was very gratifying.

The 302–940 kit and its other versions at under $250 each are a remarkable bargain, and well worth any audiophile's serious consideration. Parts Express has put together a high-quality product offering which makes an eloquent case for those audiophiles who like to have a hand in acquiring their sound systems.

This pair could be an excellent part of a beginner's system, a dorm-room setup, improved sound for a computer, or as part of a home theater expansion. A great value that would be hard to beat at the much higher prices at a local mall.

I re-wrapped the cabinets in the soft protective plastic they arrived in, and re-packed them for shipment to Joe D'Appolito's lab for measurement, and final auditioning. His findings, along with Dennis Colin's listening critique, will be published next month. ***aX***

PHOTO 7: USE PLENTY OF LIGHT TO INSTALL THE FOAM DAMPING AND THE CROSSOVER BOARDS.

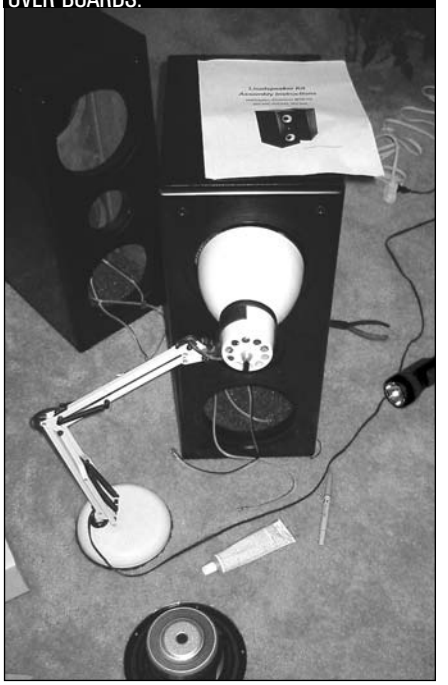
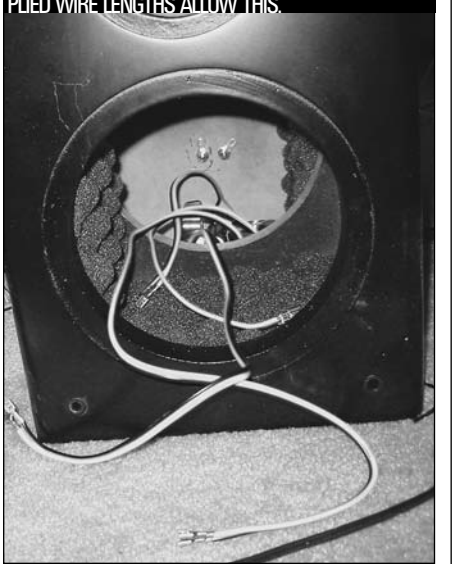
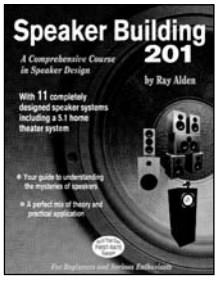


PHOTO 8: THE MANUAL ADVISES PUTTING THE CROSSOVER NEAR THE REAR. I SUGGEST IT WOULD BE EASIER TO LOCATE IT NEAR THE FRONT. THE SUPPLIED WIRE LENGTHS ALLOW THIS.



Xpress Mail

Back



RULE OF THUMB

Congratulations to Ray Alden on his new book, *Speaker Building 201* (see review, last issue). The excerpt looks to be masterfully lucid and concise.

I have one small point to make. Unless I am mistaken, the rule that governs the motion of a current-carrying conductor in a magnetic field is Fleming's Left-hand rule for motors and not the "right-hand rule of vector geometry" as stated in the section dealing with "B—the Gap" in the Oct. '04 issue (p. 23).

Not that it makes a lot of difference to us audiophiles since we don't spend much time listening to DC!

I was educated in England, and as my physics teacher (Mr. Williams from Wales) would say, "The way to remember which hand to use is that motors (automobiles) drive on the left, see?" Of course, most of the rest of the world (including Ray in the US) would need to think up a different way of remembering which hand to use.

I look forward to reading the book.

Andre Routh
Medford, N.J.

Ray Alden responds:

To start with, I thank Andre for his kind words and the gentlemanly way in which he posed his question. I will say more about this at the end of my letter, but first respond to the question at hand.

Initially, I was puzzled by Andre's mention of Fleming's left-hand rule; quite frankly, I was not familiar with it. I poked around some typical American physics books and found references to two different right-hand rules, each used in different contexts. The first was used to determine the direction of a magnetic field produced by a current flowing through a wire. If you wrap your right hand around a wire with your thumb pointing in the direction of the current flow, your fingers will point to the direction of the magnetic field.

However, the rule to which both Andre and I were referring applies to the force produced in motor-like situations. This is the second "right-hand rule" often referenced in American textbooks. It uses three fingers: the thumb, the forefinger, and the middle finger, each placed at right angles to one another.

If the forefinger points to the direction of the magnetic field into which the wire is immersed, and the middle finger to the direction of the current flow in the wire, then the thumb points to the direction of the force applied on the wire. I then discovered references to Fleming's left-hand rule on the Internet and found it does the exact same thing by reversing the alignment of the forefinger and middle finger.

The resulting thumb direction is exactly the same in each case. Andre's mention of driving on the left side in England as opposed to the right side here reminded me of the famous rift caused when differential calculus was simultaneously developed by Isaac Newton in England and Leibniz in Europe. This resulted in the use of one set of symbols for calculus in continental Europe while for centuries England used another. No matter which symbols were used, the mathematics that developed in both cases yielded the same results.

Truthfully, when I received the October *audioXpress* to find a condensed and excerpted first chapter of my book, I was both surprised and a bit apprehensive. After all, readers of *audioXpress* are more sophisticated than the beginners and intermediates for which the first chapter of the book was intended. More complex ideas—for example, the physics underlying a driver's interaction with a box, the use of multiple drivers, or the Bullock-Gonzalez modifications to crossover mathematics—appear in later chapters. I have seen vitriolic letters in both the old *Speaker Builder* and new *audioXpress* magazines from self-appointed experts who prefer to tear down rather than pose meaningful questions. Thankfully there are more readers like Andre, who have the humility to say, "I might be mistaken." Such words encourage dialog and inquiry.

OPTO-ISOLATORS

In reference to Jenoë Keceli's "Audio-Optical Isolation Amp" (a*X* 10/04), I would like to point out that the competitive product from Agilent, the HCNR-200 and HCNR-201, also perform quite well as analog opto-isolators. Agilent and Vishay have two good application notes on high-performance opto-isolators: AN1357 at http://cp.literature.agilent.com/litweb/pdf/5989-0804EN.pdf, and Vishay's AN50 (http://www.vishay.com/docs/83708/83708.pdf). The latter is a good description of unipolar and bipolar applications.

I have found that the linearity of the Agilent devices exceeds the spec cited in their term sheet.

Jack Walton
Short Hills, N.J.

Jenoë Keceli responds:

Thanks for your comment. The home pages you

cited can be very useful for further development of the optical isolation amplifier. You can also find applications for differential/balanced amplifiers on the above-mentioned sites. For practical reasons, I used Vishay when I was building the amplifier, but it would be a good idea and experience to try the Agilent device as well. Thanks for your suggestion.

I found Jenoë Keceli's "Audio-Optical Isolation Amp" with the IL300 linear optocouplers (*aX* 10/04, p. 56) quite interesting. However, I was surprised at the selection of the LM358 op amps for the audio circuitry. Low-performance single-supply op amps such as the LM324 and LM358 are just not suitable for audio use. If you inspect the data sheets, you won't find any specs related to input noise, distortion, slew rate, gain-bandwidth, and so on. The voltage follower pulse response graph on the data sheet shows the LM358 to be severely slew-rate-limited. There are not even any audio circuits in the application examples. That tells you something.

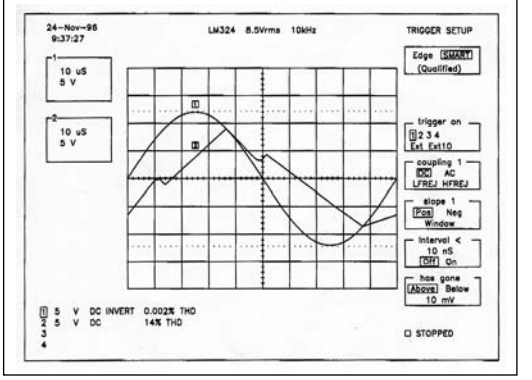
I performed some breadboard tests with one section of an LM358 configured as a unity-gain inverter into a 2k load. While the circuit does meet the 0.05% THD listed by Mr. Keceli at 1kHz and 775mV, the input signal level had to be reduced below 100mV at 10kHz in order not to exceed 1% THD at the output. With a 1V RMS 10kHz sine wave at the input, the output THD was 11%. Viewed on an oscilloscope, the output waveform was triangular at the top, fishhook-shaped at the bottom, and had spurious oscillations at zero crossing. At higher frequencies and levels the output waveform developed a "scoop-out" on the positive half-cycles.

The LM358 also sounds terrible in audio use. Given the availability and low price of pin-compatible dual devices with vastly better performance, the LM358 is not even cost-effective.

Even the mass-market consumer audio favorite, the NJR4560, would be better. However, as a minimum, I suggest the NE5532A or TL072A. These would drop right into the PC board. There are, of course, even higher-performance modern dual op amps specifically designed for audio use. Search the websites of TI/Burr-Brown and Analog Devices for suitable candidates.

Charles Hansen
Ocean, N.J.

FIGURE 1: LM324/LM358 WAVEFORM DISTORTION.



Jenoë Keceli responds:

Thanks for your interest in my article. I can say that there are, in fact, better Ops than the LM358 (but they might not perform well at low-supply voltage). I tested some for the article, and the results showed that their application at low-supply voltage is strongly limited, especially at higher gain. (This does not show

at unity gain you refer to in your letter).

Because my aim was to show how to build a simple isolation amplifier, I opted for the LM358 because of its good performance (*Fig. 1*) at low-supply voltage. (This is readily available in my country—Hungary.) The IC concerned is meant to be used at low-supply voltage. The equipment that the article refers to is not designed for the transmission of 1V RMS.

You could achieve better results with other components, higher supply voltage, and perhaps a full-balanced design, but this would result in another amplifier.

FILAMENT POWER

A.J. van Doorn's latest piece, "Filament and High Voltage Power" (*aX* 9/04, p. 26), contained a marvelous and provocative bit about "compound (power) control" for filament power. I was a bit skeptical when I saw it, since regulating power precisely entails the use of a multiplier, or some comparable nonlinear element. And indeed the LM317-based circuit shown does not regulate perfectly, but my, how well it does within its constraints! Analysis reveals that it is synthesizing a resistance matched to that of the nominal filament R, in series with a doubled nominal voltage. It is remarkable how well this works, indeed well enough to fairly trounce more complex alternatives. Delightful!

Another wise remark suggests placing a positive bias between heater and cathode in low-level circuits, which is rarely discussed.

Brad Wood
swamipoin@yahoo.com

A.J. van Doorn responds:

Thank you for the attention and your kind response to my article. Indeed you can control power very well, such as with the power control chip p/n EL4452C available at www.elantec.com. Putting filaments at a positive DC voltage is an old habit that can help if DC for the filaments is too cumbersome.

As you know, voltage feedback produces a low source impedance, and current feedback leads to a high impedance. With a combination of both current and voltage feedback, you can get almost any desired source inner resistance. I experimented a bit with the TDA 7294, a wonderful 100W audio power chip.

With voltage feedback only, the speaker experiences a very low source resistance. With both voltage and current feedback, you can simulate an inner resistance as for a triode amp, which will sound better.

The famous speaker damping factor is a joke. My Tannoys have a 6Ω impedance and a low DC resistance of about 2Ω, which is always in series with the speaker cable, so even at zero source impedance, the damping factor isn't that low.

If you like, just put a small 220mΩ resistor in the speaker return cable and play with current feedback for the best sound. Looking at the A15 tube amp schematics in *aX* 9/01 ("Glass Shard," p. 93), you'll see a potentiometer named damping control, which controls both current and voltage feedback, as well as the overall gain.

MORE SUB DETAILS

The Tuba 18 Sub (Jan. '05 *aX*, p. 12) is a project I have been waiting for, but the drawings for the inside panels "A, B, C, D, E, F, G," and so on are incomplete! The Figure 2 drawing has only partial information as to the dimensions.

I would like to build this sub, but without complete panel dimensions I will be unable to do so. Could you please provide me with a source for the missing dimensions?

Bill Hofmeier
billhof@citlink.net

Bill Fitzmaurice responds:

Quoting from the article:
"To start construction, cut out the two sides. Draw

the locations of all the intersecting parts (Fig. 2) on them, starting with the front, top, back, and bottom, and from there the horn plates, starting at the junction of plate G where it joints to the front. The internal measurements shown are the pathway clearances at each horn bend; the panel dimensions will depend on the actual measured thickness of the material used, as there is no such thing as ½″ plywood that actually measures ½″ thick."

Since the parts are installed sequentially with each part butting into the one before it, the actual dimensions depend on the materials used, and are directly measured off the layout drawn on the cabinet side as each is installed. Otherwise, they could be either too long to fit or too short, leaving gaps between them. The dimensions listed are all that are required to build the box. They are, in fact, the only calculated dimensions that I used to build the original.

I have been a fan of horns since the original Klipschorn and more than once seriously considered building one. Size and placement requirements have deterred me, however. Thus, I found Bill Fitzmaurice's Tuba 18 extremely enchanting. I most definitely will build one, and have already ordered the driver from MCM. I do, however, have a few questions about the unit's technical details:

1. What is the design frequency of the horn? From the response chart, I would guess somewhere near 40Hz, as peak response is at 50.
2. What flare rate (conical, exponential, tractrix) did you choose and why?
3. What do you consider the throat area? It appears that the driver fires directly into side "E," so that the throat area would be considered the cross-sectional horn area somewhere near the midpoint of baffle "A." This strikes me as a really neat way to mount the driver relative to the horn. I can understand the necessity of this arrangement given the size restrictions, but are you aware of any particular advantages/disadvantages of this geometry?
4. With respect to placement, you speak of "facing into a room corner." I would think that you would want the mouth facing out into the room so that the floor and walls form a continuation of the mouth (à la Klipschorn), but looking back at your Tuba 24 article (April '04 *aX*), I see you specifically talk about firing it into the corner. I suppose that if you placed it a specific distance from the corner, you could get one more "turn" in your horn.

Could you be specific about where you place the mouth relative to the walls and floor of the corner? Maybe even an illustration? Also, I would love to hear why and how this works to extend the horn.

Thanks for designing and sharing this great little horn.

Mark Parker
dmtp@adelphia.net

Bill Fitzmaurice responds:

The horn pathway is about 7′, so the nominal horn fc when freestanding would be about 40Hz, though corner placement adds another foot and a half of effective path length, so 35Hz would be more like it. The flare is a series of conics, chosen to minimize the cabinet size. The throat technically begins at the point where the cone is no longer firing into the area between parts A and E; the horn pathway prior to that is technically part of the driver front chamber.

With a bass horn there is no disadvantage to the driver mounting topology, as the long wavelengths being passed are omnidirectional. Likewise, frequencies in the passband are too long for the pathway differential between the leading and trailing edges of the cone to cause any phase-related response glitches.

Having the box facing into a corner duplicates a KHorn alignment, as the final section of the horn pathway and mouth is formed by the outer walls of the cabinet and the room walls. I tried a triangular top à la the Klipsch, which didn't improve response so I didn't use it. My own sub is placed kitty-corner, spaced away from the walls on either side by only the ¾″ thickness of the baseboard moldings.

OVER THE YEARS

I am very pleased with the transition to *audioXpress* from *Glass Audio*. I find the other articles about speakers and solid-state projects interesting and helpful.

I would like to thank the authors who have inspired me to build the equipment that I use today. I have built the headphone amp found in the trial (0/88) issue of *Glass Audio*. This project by Becker and Oberesch has become my reference system.

I've built the "Peak Reading Level Meter Using Indicator Tubes" (2/95) by Jukka Tolonen. Gary McClellan authored an article, "EZ Start Power Controller" (3/96) that I use on my older tube amps

to help preserve the B+ filter caps and expensive power tubes. Eric Barbour's article, "Improving the SV811-10 Amplifier," (5/97) was a follow-up that includes modifications on an earlier project. This was my first single-ended tube project and proved to be a good match for my Klipsch speakers.

My favorite amp is Joseph Norwood Still's "40W Triode/60W Ultralinear Amp" (2/98), which I keep set up most of the time in my listening room. I also enjoy his later articles in which he experiments with different designs and variations on this amp. I plan on utilizing some of these tweaks on the amp I built from his first article. I have also purchased all the parts to build Mr. Still's "High Quality Control Unit" (*aX*, March '01) and plan to build this preamp in the near future.

Peter Millet submitted an article on speakers in *audioXpress* and briefly mentioned his 813 transmitter tube amp. He supplied schematics and websites for this amp. I am now acquiring parts to build this, too.

I would like to thank all the authors that take time to document their work and submit them in writing for publication. I admire and respect their expertise and dedication that brings us the articles found every month in *audioXpress*.

Jim Dungan
Port Neches, Tex.

TRANSFORMER POWER

The power transformer Rick Spencer specified ("A Mini SE Amp," *aX* April '04, p. 6) handles 269mA of current draw, but the choke specified is rated at 150mA. I have a number of possible transformers I could use, but really need to know what the current need actually is.

Bruce Brown
Tuninfork@aol.com

Rick Spencer responds:

I thank Mr. Bruce Brown for his interest in my project. I am certain that once he has built the amplifier, he will be surprised and pleased with the quality of sound for such a small investment.

You are correct regarding the quoted specs of the transformer and the choke. When selecting the B+ transformer for any project, I like to use one with a power rating that is approximately 25 to 30 percent greater than the circuit will normally demand. This will prevent any overheating of the windings and will, of course, extend the life of the tranny itself.

Sometimes it is difficult to find HV transformers with the proper rating, especially when constructing amplifiers that are capable of very high power output. Most high power units require separate transformers to supply the various branches of the circuit, such as B+, heaters, or filaments, and even the bias voltage. The same is true concerning the size of the choke.

If you wish to use choke filtering for all the B+ circuit, then choose a choke that can handle all of the current required throughout the entire circuit. Some of these larger value chokes are quite heavy and have dimensions that are almost the same as some power transformers. On the other hand, some may wish to filter the B+ supplied to the driver tubes only, which may be accomplished with a smaller type of choke.

The transformer I chose for the mini SE amp is perfect for the application. Concerning the load on the B+, if you add up the total draw of all the tubes, you will find that the idle current is only around 110mA and not much more at maximum output. The Hammond #261M6 is just loafing along at this rating while being used in the choke input filter configuration. The choke is rated for 150mA while maintaining its value of 5H.

This choke, like most others, will have a little less inductance when operated at a DC current higher than recommended and will show a little more when used at a lower current. When I pushed my amp hard during testing (maximum power for periods of ten minutes or more), neither the transformer nor the choke became more than slightly warm to the touch.

I think that all Hammond products are rated very conservatively, and, from my experience, I have never had one roll over and quit! So, as you can see, if you build the amp with a transformer and choke within these stated parameters, you can't go wrong. Just remember to use a good choke for filtering in your power supply, because even with extra large capacitors, some noise will sneak right on by and into your amp circuits, and it will take a good choke to keep your B+ clean. If you still desire a "beefier" choke, then I suggest the Hammond #159S, but you will definitely need a larger chassis than I used.

You stated that you had some transformers on hand. Just remember to keep the B+ that hits the plates of the 12L6s within the 200V range. The Hammond transformer and choke combo that I used gave me the required voltage without requiring any dropping resistors, so perhaps you will be just as fortunate.

Good luck, and enjoy building your version of this neat little amplifier. And, please, write back to *audioXpress* when you are finished, and let everyone know how it turned out! ***aX***

Back

VENDORS

Mic and phono preamps, meters, filters, etc. all US built. **TDL Technology, Inc., 505-382-3173 www.zianet.com/tdl**

Principles of Power, tube audio books, kits, FAQ. **www.londonpower.com**

DIY Starter Kits
www.borbelyaudio.com
Selected Borbely Audio kits in North America:
www.audiokits.com
Selected Borbely Audio kits in Japan:
http://homepage3.nifty.com/sk-audio/

www.stillaudio.com
18 watt 0.6% distortion (no loop feedback) single-ended amplifier

AD INDEX

Yard Sale

FOR SALE

18W per channel Single-Ended Class A Ultra-linear with volume control. Can be driven by CD. $1700 plus shipping.

LMS (loudspeaker measuring system) version 2.51 including test cables. $1800 plus shipping.

(250) 497-2021

Liberty Instruments Audiosuite

$275.00 plus shipping

John (508) 835-4490

"Yard Sale" is published in each issue of *aX*. For guidelines on how subscribers can publish their free ad, see our website.

solid state

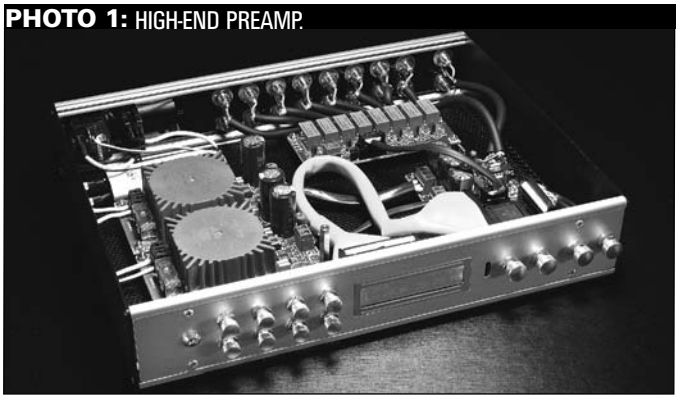
High-End-Preamp, Part 2

By Benjamin Hinrichs

Back

The modular design and flexibility of the microcontroller program allow the preamplifier to be built exactly according to the individual requirements of its user.

PHOTO 1: HIGH-END PREAMP.



The multitude of options for constructing your "personalized" high-end preamplifier means that you must read this second part of the article before you order the parts and take up your soldering iron. This is because your requirements determine how many circuit boards of which type you will need, as well as which components you will need. You can connect several volume controls in parallel as illustrated in *Fig. 1*, in order to adjust more than two channels at the same time. If you do this, you must interconnect the CS, Mute, SCLK, +5V, and Ground terminals of the circuit boards. SADATO from the first circuit board is connected to SDATI of the next circuit board, and so on. This causes the volume level setting to be communicated to all of the circuit boards.

You can choose from traditional audio signal switching, switching the ground leads together with the signal leads, or switching balanced signals. If you know how, you can also give the microcontroller additional functionality or modify the details in the software, such as the designations shown on the display.

POPULATING THE BOARD

What we need now are a few instructions for populating the main preamplifier board. This is because the circuit board shown in *Fig. 2* is quite compact. The closely spaced, narrow tracks are not particularly forgiving of mistakes, so the components should be fitted properly the first time, and solder splashes and overheated joints should be assiduously avoided.

On the main circuit board, only JP1 and JP3 can be implemented as jumpers. JP2 (directly underneath IC2) is a wire bridge. There is another wire bridge at the inner edge of K5, which should be fitted immediately: done is done!

Now you should consider whether the programming and/or paralleling

FIGURE 1: CONNECTING SEVERAL VOLUME CONTROLS IN PARALLEL.

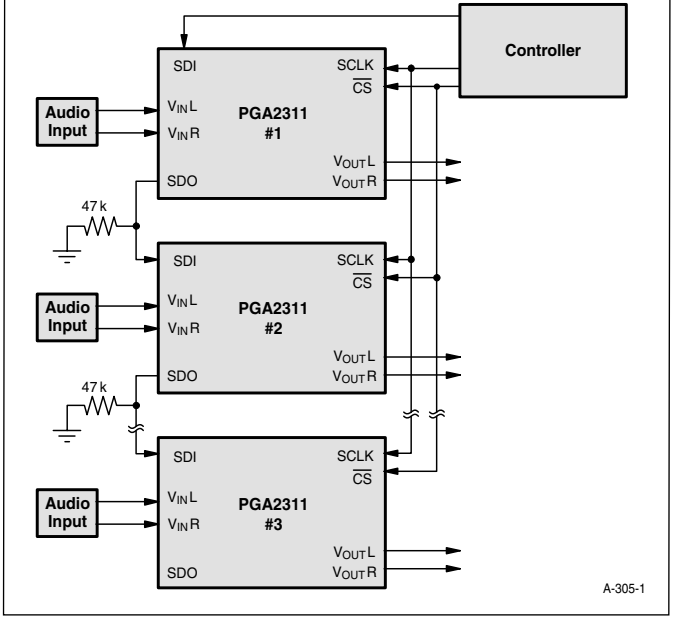


FIGURE 2: COMPONENT LAYOUT OF THE TIGHTLY ARRANGED BUT SINGLE-SIDED MAIN CIRCUIT BOARD.

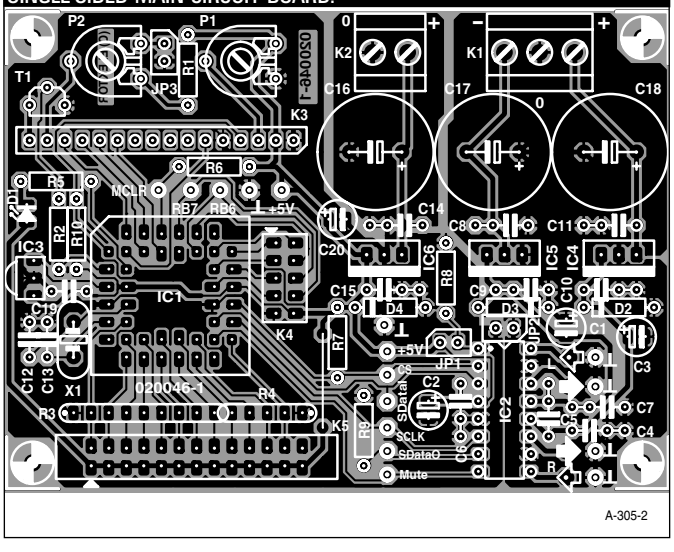
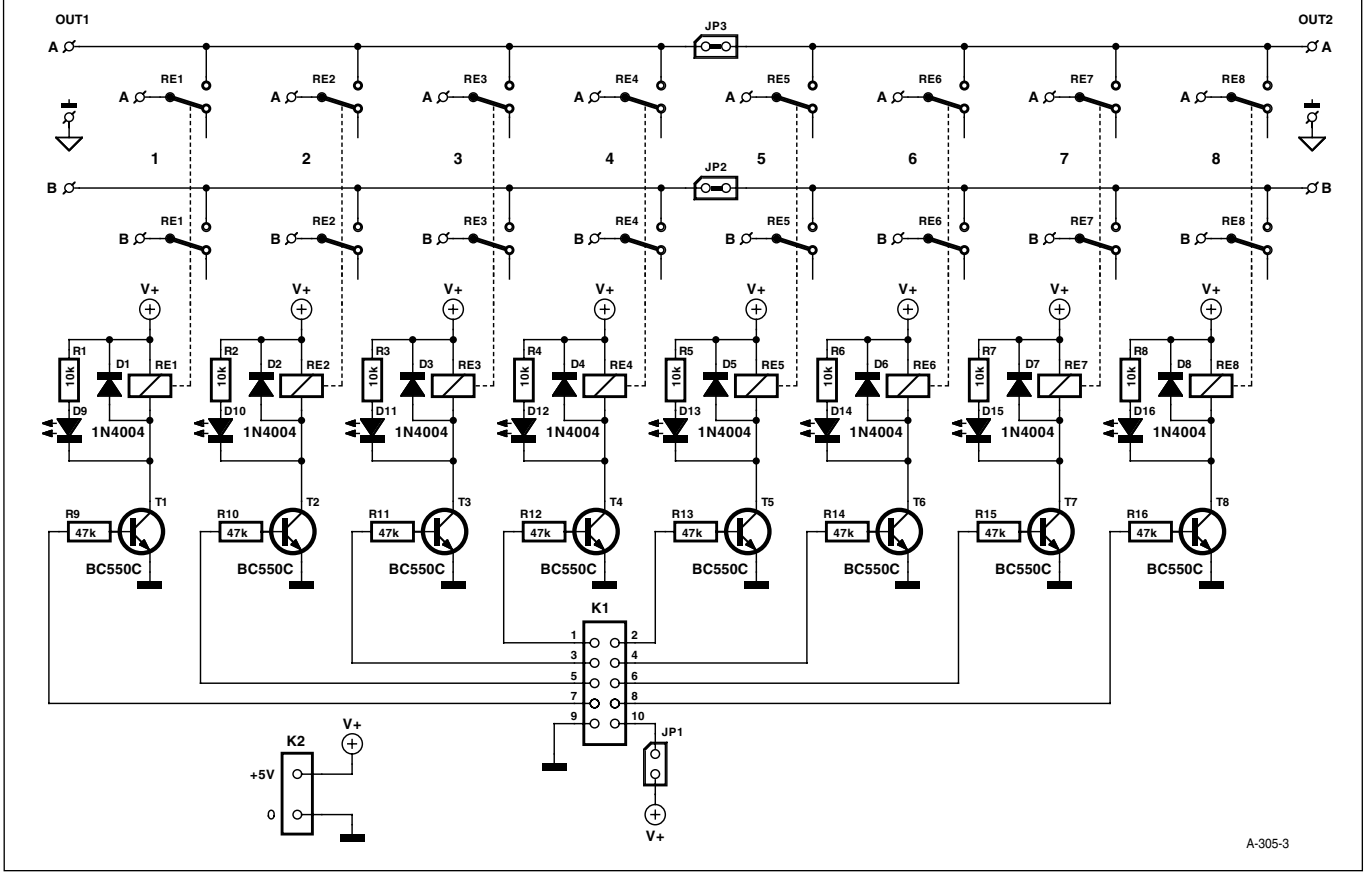


FIGURE 3: THE RELAY CIRCUIT ALSO ALLOWS GROUND ISOLATION AND CONNECTING BALANCED SIGNAL SOURCES.



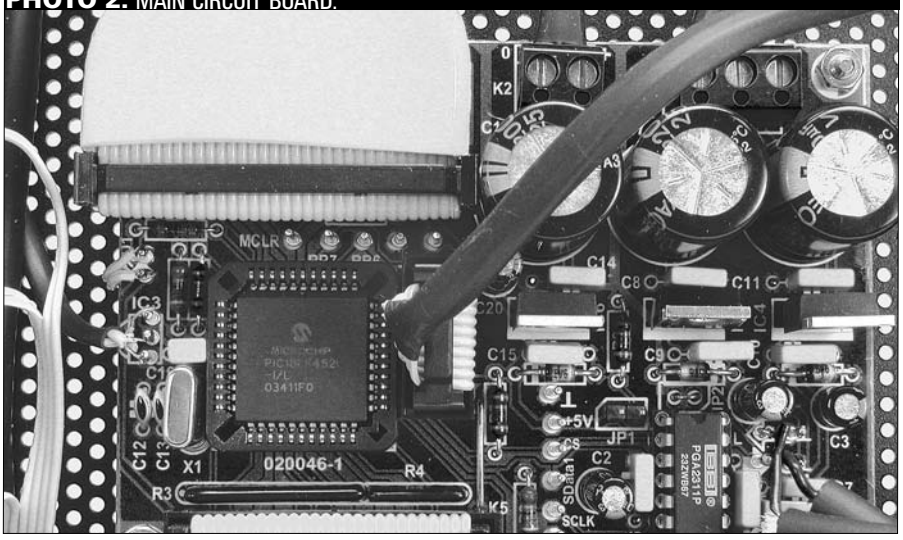
connections are necessary. If you arrive at a positive conclusion after studying the two text boxes, then you must fit the associated connectors (refer to the components list) and the solder pins for the audio signals. Despite all the claims to the contrary, it certainly makes sense to fit the solder pins before the other components. Generally speaking, they need a bit of coaxing, and sometimes they can only be pressed into the holes using (brute) force. Besides this, a 1.3mm diameter silver-plated steel pin requires a considerable amount of heat to achieve a reliable bond with the solder. And if you find a solder pin standing at an angle to the board, you can start all over again. This can be the kiss of death for small, sensitive components that have already been fitted in the immediate vicinity.

If you use a decent stuffing frame, it doesn't matter very much if the low-profile components (resistors, capacitors, and diodes) are only fitted afterwards. As always, proper polarization is important for some of the components. Besides the diodes and small electrolytic capacitors, this also includes the resistor arrays and the PLCC socket, which is beveled on one corner. This marking must match the component overlay on the circuit board. Now is the right time to **not** fit LED D1 if you want to use it as a pilot light on the front panel. This LED, along with IR receiver IC3, should only be soldered in place after the circuit board has been securely fitted in the enclosure and the two components have been fastened in holes drilled in the front panel. If your enclosure layout does not permit the circuit board to be placed directly behind the front panel and you thus must fit the IR receiver off the board, a screened connection cable is mandatory.

The fixed voltage regulators IC4–IC6, which should be soldered in place after pin headers K3–K5 and the high-quality DIL socket for the PGA2311 have been fitted, do not require heatsinks. The soldering work is completed with the fitting of the three large electrolytic capacitors and the two circuit-board terminal strips (K1 and K2).

If you wish to control several volume control boards in parallel, fully populate one main circuit board and fit only the portion with the PGA2311 (including

PHOTO 2: MAIN CIRCUIT BOARD.



the peripheral circuitry and the voltage regulator circuitry) on the other boards.

The system has been tested using a single controller board and two parallel circuit boards. All of the necessary connection points are located in a row on the circuit board.

CHANNEL SWITCHING

The input channels are switched on the relay board. The circuit shown in *Fig. 3* allows two different wiring options.

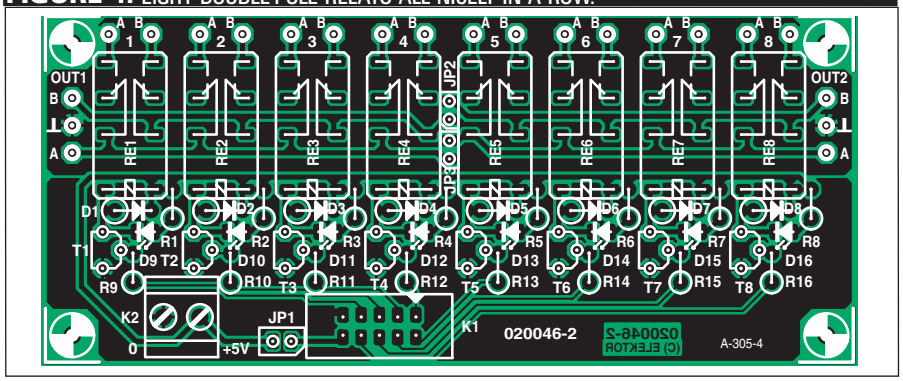
With the **traditional option,** all eight inputs from the Cinch sockets are used, with a single output to the main circuit board. This requires JP2 and JP3 to be fitted. Each relay is then responsible for one stereo channel. If standard twin screened audio cable is used, one channel is connected to A and the other to B, and the screen braid is soldered to the bottom of the circuit board.

Ground isolation, which may be desired to improve channel separation or avoid interconnecting the grounds of different items of equipment connected to the preamplifier, can be achieved by switching not only the signal lines, but also the associated ground potentials. In this case, for each channel the "live" lead is soldered to A and the ground lead is soldered to B. In addition, jumpers JP2 and JP3 are left open. In this mode, each pair of relays (RE1 & RE5, RE2 & RE6, RE3 & RE7, and RE4 & RE8) belongs to a single signal source. This yields four stereo inputs and two sets of outputs (OUT1A & OUT1B and OUT2A & OUT2B). Here again A corresponds to the "live" lead and B to ground. The only other thing you have to do is to change the Input Type option in the Set-up menu to "Double."

In order to handle **balanced signals**, we simply switch two relays simultaneously. In this mode, you also have to add a second, parallel volume control and then connect the balanced signals to one of the input channels (such as Left In) on both volume controls. This works well—quite well, in fact.

Fujitsu (formerly Takamisawa) type RY-5W-K relays are used here. Naturally, other pin-compatible types could also be used, but the specified line has excellent characteristics and is available for less than 2 euros (approx. £1.40, $2.30), for example from Conrad Electronic (www.conrad.de). This is a plastic-encapsulated relay, which prevents any dust from accumulating inside. The contacts are made from a silver–palladium alloy with supplementary gold plating.

FIGURE 4: EIGHT DOUBLE-POLE RELAYS ALL NICELY IN A ROW.



Power for the relay board is taken from the main circuit board via connector K1 (which requires jumper JP1 to be fitted). If you wish to treat the channel switching board to its own power supply or you want to use this board in a different project, you can also connect a 5V supply to the K2 terminals.

In order to avoid overloading the microcontroller outputs, transistors with base resistors are used as relay drivers. Diodes D1–D8 act as freewheeling diodes, while LEDs D9–D16 provide a visual indication of the relay states. The LEDs do not necessarily have to be fitted to the circuit board (*Fig. 4*); they can of course be fitted to the front panel instead if you want to do without the display, or they can simply be omitted (along with their series resistors).

JP1 can be implemented as a jumper, but JP2 and JP3 should be implemented as wire bridges. Otherwise there's not anything particularly remarkable about populating the circuit board, particularly if you use the components specified in the components list and everything fits perfectly.

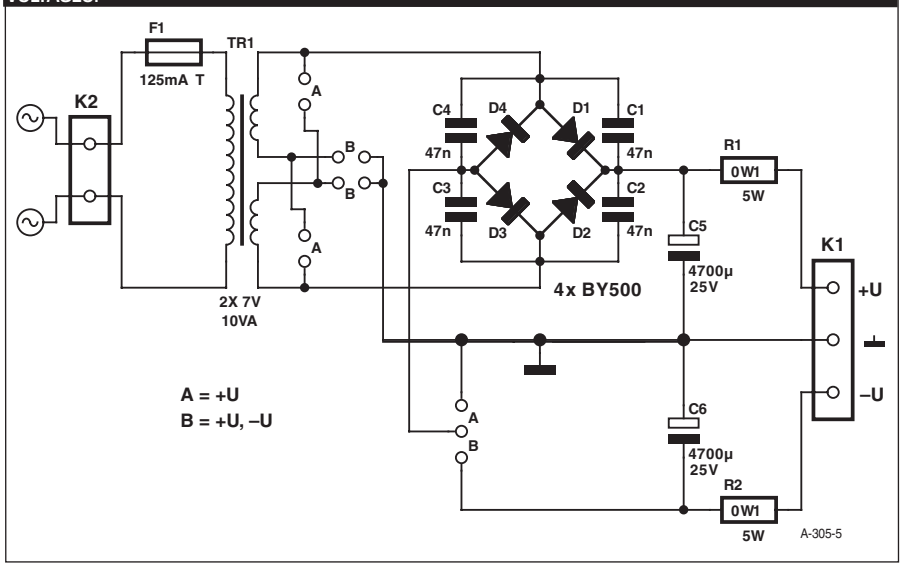
STABLE SUPPLY

A large part of the stabilization, decoupling, and hum suppression for the supply voltages takes place locally on the main circuit board, for both the digital portion (+5V) and the analogue portion (5V). The only other thing you need is a sufficiently stable source of symmetrical DC voltages (which shouldn't be a problem with the low current consumption). This provides an opportunity to use a general-purpose low-power mains power supply (*Fig. 5*).

The power supply is designed to use Talema 10-VA encapsulated low-profile toroidal transformers (see reference 2), which can be obtained from DigiKey (among others). Particularly for audio applications, toroidal transformers are preferable to other types of transformers due to their low stray fields.

The power supply circuit board is gen-

FIGURE 5: THE MAINS POWER SUPPLY CAN PROVIDE UNIPOLAR OR SYMMETRIC OUTPUT VOLTAGES.



eral-purpose because it can be configured to provide either an asymmetric supply voltage (V+ and ground) or symmetrical supply voltages (V+, ground, and V–) by simply fitting either wire bridges "A" (for asymmetric output) or "B" (for symmetric output). Naturally, C6 and R2 are not needed if it is configured to provide an asymmetric supply voltage.

The output resistors (R1 & R2) should be wire-wound types, since this further reduces the hum voltage. Fast diodes with a recovery time of less than 200ns are used as rectifier diodes (D1–D4)—for comparison, the recovery time of standard 1N4004 diodes is 1.2µs, which is more than six times as long. If you wish to use ultrafast diodes (< 50ns), you are of course free to do so.

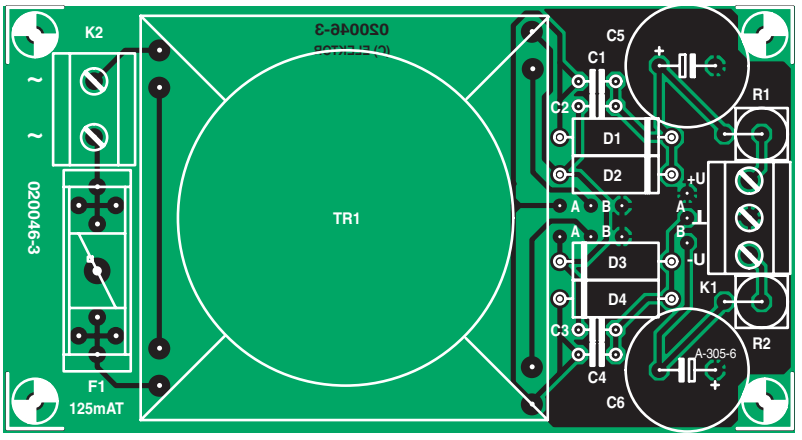
Small foil capacitors are connected in parallel with the rectifier diodes to effectively suppress high-frequency noise, which primarily arises from diode switching transients. Each circuit board is protected by a slow-blow fuse rated at 125mA.

Fitting the components to this board (*Fig. 6*) is also about as simple as could be imagined. First solder the wire bridges in place (one circuit board with the three "A" wire bridges and a second one with the three "B" wire bridges). Then fit the remainder of the components in any desired order. Mount the series resistors vertically.

TESTING, TESTING, TESTING. . . AND INTO THE BOX

Before inserting the microcontroller and the PGA2311 into their sockets and connecting the display, you should painstakingly check all the boards for incorrectly fitted components, dubious solder joints and solder bridges. The first test should be performed with the boards outside of the enclosure. To hold the boards securely in place for testing,

FIGURE 6: THE POWER SUPPLY BOARD SHOULD BE FITTED AS FAR AWAY AS POSSIBLE FROM THE MAIN CIRCUIT BOARD.



screw them to the workbench using standoffs under each of the boards.

First connect the power supply boards and check the voltages at their outputs. Due to the high open-circuit voltage and the combination of rectification and filtering, the measured voltage will naturally be somewhat higher than the 7V specified for the transformer.

Next, connect all of the boards together and switch on the power.

Nothing will happen, but the following voltages should be present at the appropriate IC pins and the display module connector:

+5V:
K3–2, IC1–12, IC1–35, IC2–4, IC2–12

–5V: IC2–13

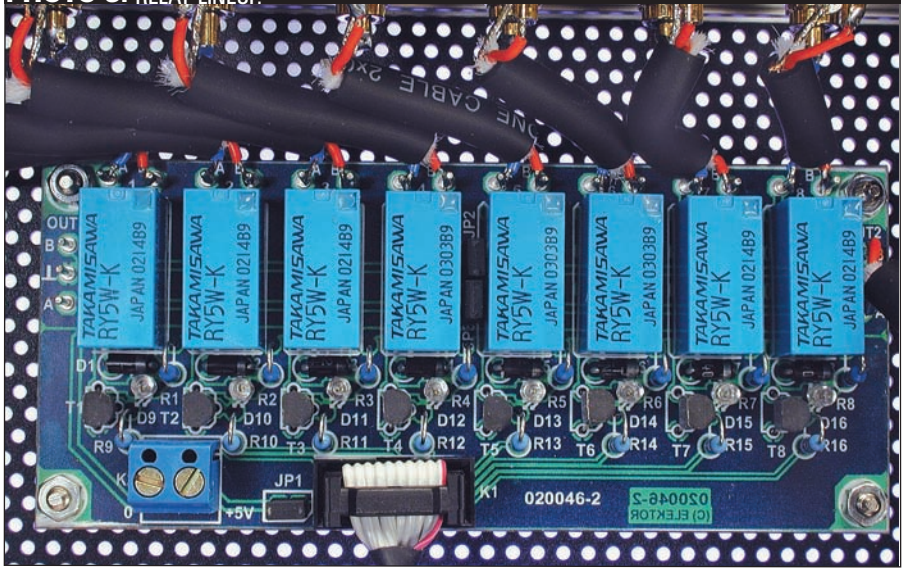
If everything is OK, wait a few minutes to allow the capacitors to discharge, and then insert the ICs and connect the display.

The display, the microcontroller, and the volume control IC are highly sensitive to static electricity. As all of these components are not exactly inexpensive, you should work at a suitably earthed workstation.

Figure 7 gives a deep insight into the laboratory prototype of the digitally controlled high-end preamp. The mains filter was fitted because it was "already on hand," but it is fully adequate. The power supply boards and the main board are separated as far as possible. Nevertheless, the FFT analysis (see Part 1) shows that induced 50Hz noise is responsible for the majority of the distortion. If you want to improve matters and are not afraid of a bit of effort, you can fit the sensitive audio portion into a suitable tinplate enclosure.

At the rear, besides the mains input

PHOTO 3: RELAY LINEUP.



COMPONENTS LIST MAIN BOARD

Resistors
R1 = 27Ω
R2, R9 = 47kΩ
R3 = 8-way 10kΩ SIL array
R4 = 4-way 10 kΩ SIL array
R5 = 1kΩ5
R6, R7, R8, R10 = 10kΩ
P1 = 10kΩ preset
P2 = 100Ω preset

Capacitors
C1, C2, C3 ,C20 = 10µF 25V radial
C4–C11, C14, C15 = 100nF
C12, C13 = 27pF
C16, C17, C18 = 2200µF 25V radial
C19 = 100nF ceramic, lead pitch 5mm

Semiconductors
D1 = low-current LED (+ 2-way pinheader)
D2, D3, D4 = zener diode 5V6, 1.3W
IC1 = PIC18LF452-I/L (PLCC). Blank ICs: Farnell # 400-9654. Programmed ICs: order code 020046-41
IC2 = PGA2311PA from Texas Instruments/Burr-Brown or CS3310 from Cirrus Logic (Crystal)
IC3 = SFH5110 (TSOP1836) (+ 3-way pinheader)
IC4, IC6 = 7805
IC5 = 7905
T1 = BC550C

Miscellaneous
JP1, JP3 = 2-way pinheader + jumper
JP2 = wire link
K1 = 3-way PCB terminal block, lead pitch 5mm
K2 = = 2-way PCB terminal block, lead pitch 5mm
K3 = 16-way pinheader
K4 = 10-way boxheader, vertical
K5 = 26-way boxheader, vertical
X1 = 10MHz quartz crystal
12 solder pins
External parts:
LCD, 2 ×16 characters with backlight
12 pushbuttons for chassis mounting
2 Cinch sockets for chassis mounting, isolated and gold-plated
Mains on/off switch, chassis mount
IEC mains appliance socket, chassis mount
Disk, hex and source code files, order code 020046-11 or Free Download.

RELAY BOARD

Resistors
R1–8 = 10kΩ
R9–R16 = 47kΩ

Semiconductors
D1–D8 = 1N4004
D9–D16 = low-current LED
T1–T8 = BC550C

Miscellaneous
JP1 = 2- way pinheader + jumper
JP2, JP3 = wire link
K1 = 10-way boxheader, vertical
K2 = 2-way PCB terminal block, lead pitch 5mm
RE1–RE8 = RY5W-K (Takamisawa), Conrad Electronics # 502852 (5V/167Ω)
16 Cinch sockets for chassis mounting, isolated and gold-plated

COMPONENTS LIST PSU ASYMMETRICAL

Resistors
R1 = 0Ω1 5W

Capacitors
C1–C4 = 47nF, lead pitch 5mm
C5 = 4700µF 25V radial, max. dia. 16mm

Semiconductors
D1–D4 = BY500-200

Miscellaneous
F1 = fuse, 125mA, slow, with PCB mount holder
K1 = 2-way PCB terminal block, lead pitch 5mm
K2 = 2-way PCB terminal block, lead pitch 7.5mm
TR1 = mains transformer, 2 × 7V 10VA, low profile,e.g., Talema 70040 (RSComponents, Digikey)

Fit wire links "A"

Do not mount R2, C6

SYMMETRICAL

Resistors
R1, R2 = 0Ω1 5W

Capacitors
C1–C4 = 47nF
C5, C6 = 4700µF 25V radial, max. dia. 16mm

Semiconductors
D1–D4 = BY500-200

Miscellaneous
F1 = fuse, 125mA, slow, with PCB mount holder
K1 = 3-way PCB terminal block, lead pitch 5mm
K2 = 2-way PCB terminal block, lead pitch 7.5mm
TR1 = mains transformer, 2×7V 10VA, low profile, e.g., Talema 70040 (RS Components, Digikey)

Fit wire links 'B'

REFERENCES AND LITERATURE

1. www.fcl.fujitsu.com/en/products/relay/index.html.
2. www.talema.de
3. www.avtechpulse.com/appnote/techbrief9.
4. www.jdm.homepage.dk/newpic.htm.
5. http://jal.sourceforge.net.
6. http://groups.yahoo.com/group/jallist.

SOFTWARE MODIFICATIONS

Microcontroller programming is normally not the forté of audiophiles who have dedicated themselves to DIY construction. (Contact *Elektor Electronics*, sales@elektor-electronics.co.uk. regarding availability of microcontroller—order number 020046-41). However, anyone who has a suitable development package for the PIC18LF452-I/L will certainly not have any problems downloading the hex code for the control program into the microcontroller (item number 020046-11, available at no charge from the Free Downloads page of the *Elektor Electronics* website http://www.elektor-electronics.co.uk/dl/dl.htm). The easily constructed JDM programmer (see reference 4) is an example of a suitable programmer.

The software package also contains the source code, so interested users can get an idea of how it is constructed and/or make their own modifications (assuming they have suitable programming experience). The software is written in the JAL high-level language (see reference 5), which is available from the Internet as freeware. JAL is an excellent choice for getting started with PIC microcontroller programming, particularly since there are also very informative newsgroups on this subject (see reference 6). Nevertheless, it was necessary to make a few modifications to the JAL files in the assembly code since JAL does not provide native support for storing code in the EEPROM of the PIC18F452. All of the modified files are included in the project. If you wish to make changes to the source code, you should adjust the path names in the accompanying JAL.BAT file and use them for compiling. The software was compiled against JAL 0.4.59.Win32.

The software was entirely written by the author, except for a few functions such as driving the display, for which JAL provides native support. Functions such as the SPI interface code and acquiring RC5 signals were specifically developed for this project. Modification and/or reuse of parts of this software, including in your own projects, is explicitly desired, as long as the author is named and the software is used for non-commercial purposes (which means at home). More on this subject can be found in the file VV.JAL.

The PIC18LF452 can easily be programmed in-circuit. For serial in-circuit programming, a number of conditions must be satisfied: the PBGA2311 must be removed from its socket, and the connectors for the display and relay board must be disconnected. The programming interface comprises five connections: MCLR, RB6, RB7, +5V, and Ground. These connections are arranged in a row on the circuit board and fitted with solder pins.

socket there are two sets of eight audio inputs and two audio outputs, all using isolated gold-plated Cinch sockets. After the "official" completion of the device in the *Elektor Electronics* lab, further tinkering revealed that small RC networks on the inputs suppress induced RF noise on the cables, and that a 100Ω series resistor in each of the output lines assures the stability of the PGA2311. As shown in *Fig. 8*, these components should be soldered directly to the Cinch sockets using the shortest possible leads, or even fitted in the connecting cable, in order to maintain the highest possible degree of screening.

In contrast to the audio connections, which employ high-quality twin screened cable, the keypad (which does not have a circuit board) and the front-panel LED are connected using simple flat cable. This allows a relatively large amount of freedom in the layout of the front panel. For example, the volume buttons can be arranged in a diamond pattern (if sufficient room is available). You can spare yourself the trouble of looking for a distributor for the attractive metal buttons used for the prototype—they were personally turned by the author.

The bright red cable between the main circuit board and the LC display came from a computer shop. But be careful: many of the cables available in such shops have crossed lead pairs. In case of doubt, a "normal" flat cable is also perfectly satisfactory. ***aX***

PHOTO 4: POWER SUPPLY.