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Sound Strobe: *Speaker Tester & More*



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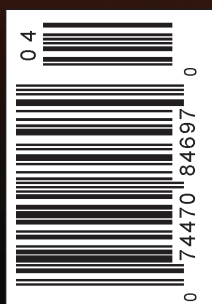


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PRODUCT REVIEW

The Sound Strobe

Discover what this sound system analyzer can do. . .
or at least some of what it can do.

By Ed Simon



I looked for Dennis Colin's picture at the Post Office, but I couldn't find it.

He had the audacity to come up with the Sound Strobe (*aX*, March '06, p. 17). When I first read the article I thought to myself, Colin (possibly with some help) came up with a new twist. The idea of using a simple 555 timer to produce a voltage spike that shoots up and down very quickly with a second or so between pulses certainly is not new.

CLICKING SOUNDS

This is known as the unit impulse. A perfect unit impulse response is a jump from 0V to some fixed level instantly and back again. There is an infinite rise time, zero time at the top, and a perfect fall time. So, of course, it only exists in theory. A practical impulse has a rise and fall time at least ten times faster than the system you are testing. The time at the full voltage should last just long enough to allow you to take good measurements.

How a system responds to this impulse can tell you just about all you need to know. If the frequency response does not go high enough, you get a rounded top instead of a sharp square edge. If the response does not go low enough, the top slopes. If the unit has too much energy storage, the pulse does not fall fast

enough. Getting the pulse size right is a bit of an art. . . until now.

The unit impulse generator is the old method we pros used to time-adjust large sound systems with audio delayed loudspeakers. If you heard two clicks you knew the delay was off. With a bit of skill you could listen until there was one click that was not "fat," and when it became neutral you knew you had the delay dialed in.



Of course, when Smaart software came out, almost everyone started using it instead. It was reasonably cheap (today you can get started for \$695 plus a computer, a mike preamp—\$100-\$600, and a microphone—\$400++) and showed you not just the unit impulse response but also let you take a reading to determine how far out you were. Like all good measurement systems, if you use them and listen, you train your ear and

can often do without the gear after a while.

When I was putting a system into the L.A. Forum, the old road dog Ted Leamy from JBL stopped by. There was an issue of how much delay to put in the side fills; he offered to set up a Smaart rig to measure it. I walked into the zone and listened, used my experience to dial in a number, and we both agreed that was close enough. Because there was only a zone of coverage where the timing would vary anyway, there being no absolute answer, good enough was the right answer. You could save a bit of money if you used the Sound Strobe to set up a delay, but you would not have all the bells and whistles of a full-blown rig.

SPEAKER DEMO

When I first got a Sound Strobe, I plugged it into the office loudspeakers (*aX* 1/07, "A Coax Horn") using a single-ended class A amplifier. After a nice clean click, I tried a few of the settings and heard nothing new. . . until I tried the "frequency" rate knob.

In my article on this loudspeaker, you may remember I had a problem measuring the low-frequency response. Although my measurement microphone is rated -1dB at 9Hz, the noise floor of my testing space rises as frequency drops

so that I was unable to get a reasonable low-frequency plot. The Sound Strobe pulse began to behave differently below a 17Hz repetition rate! This was enough evidence for me to make a few simple listening checks to tweak the speaker's position in the corner, something I could not do with my other gear.

The next day I showed the unit to a client who came in to pick up a repaired speaker, and to one of my guys. We hooked up the speaker, waited until it clicked, and demonstrated that the speaker worked. We got a nice clean click out of it until I twisted the waveform knob. At the "Exponential Pulse 830 μ S" setting, we all heard two clicks coming out of the loudspeaker, indicating the crossover point was well out of time adjustment. This loudspeaker is a widely copied product, but even with the knockoffs it still sells. Last I knew, about \$15 million a year worth of product in the pro market.

I mentioned the double click to Bill (he can hear pretty well), who was one of the guys responsible for the speaker design. His response was, "Where do I get a Sound Strobe?" He was as surprised as I was, because the normal measurements and user responses did not show this problem.

ON THE ROAD

One of the common misconceptions in audio is that the voice coil is the center of the time origin for loudspeaker driver components. Actually, it is not. Sound travels about ten times faster through a cone than through the air, which is why for time origins determinations where the forward edge of the piston couples to the air requires a rough guess. There is usually considerable argument when you try to explain this to a "True Believer," unless you can actually demonstrate this. Of course, to really do it right required complex test equipment. . . until now.

This is where that Colin fellow becomes downright dangerous. There are six settings on the "Waveform" knob, each of which does something useful. The timings he picked for the exponential decay are pretty much dead on to where they are useful. Just twiddle the knob and something



jumps out at you! Lucky guess, I suppose.

I took the tiny box with me on a few system checkouts and tuneups. After a few hours of adjusting a small church (600 seats) system, I put a few impulses through the system and it confirmed what I fixed, and what was wrong with the room was still wrong with the room. No new information, just a last-minute check to be sure I had not overlooked anything that would possibly embarrass myself.

At a bigger joint (18,000 seats) it told me the good seats were good and the bad seats were acceptable. I could hear the pulses "fatten" where the sound was not as good, but the STI meter—although it dropped—was still in the acceptable range.

SETTING THE TRAP

Although it was a great double-check box, most *aX* readers probably won't use it to check out a stadium or even an auditorium. Because I had a nice woofer

left-over, I thought I would build a pretty standard two-way bookshelf speaker to give Colin's box one last chance to show that it could not do everything.

While talking to the woofer supplier about another project, I asked for a suitable tweeter and I bought a pair based on his suggestion. This is one of the pro sound secrets: Ask the guy selling the audio gear what he recommends. If it works out, he knows what he is talking about, so you can trust him again. If it doesn't, find another place to shop.

I put the speaker parameters into one of my box programs to determine box size. I then visited a local woodworker's store and picked up some walnut sticks on sale for \$5 a board foot. I needed to cut the boards into smaller pieces consisting of a side and top length, plus allowance for planing.

After planing, edging, and ripping, I built a small sled to crosscut the pieces. Even though I had just planed the wood, I still did not cut a 45° joint and flip the board over to save wood. After I cut it to final length, the wood still cupped enough that the seam would not close nicely, so I kept all the outsides out, and the cup in. I used an old mailing tube for the port, one that fit snugly into the hole left by my 2" hole saw (**Photo 1**).

I put together an Excel spreadsheet to calculate the frequency response of a crossover and estimated what component values I would need based solely on the datasheets for the loudspeaker components. I then built some capacitor, inductor, and resistor boxes. The capacitor boxes use 1, 2, 4, and 7 μ F capacitors with a two-pole 12-position switch to give a value between 1 to 9 μ F. I used a meter to assemble reasonably accurate values out of my junkbox assortment of film capacitors.

I wound my own tapped inductors with 18-gauge wire for the inductor box. I made a form from a dowel 1" by 1½". I found that the first tap at 99 turns gave me 250 μ H. The next 45 turns brought the total inductance up to 500 μ H. Then 39 turns to 750 μ H and 33 more to 1mH. In case you want to go further, try 31, 29, 24, 22, 18, 16, and 16 turns to get you to a nicely stepped total of 2.75mH.



PHOTO 1: Test boxes to determine Sound Strobe results.

For the resistor box I used just ten 1Ω 2W film resistors and, of course, a 12-position rotary switch—single pole. I now had all the makings of a do-it-yourself adjustable crossover kit. I used to have a single box for this, but after I lent it to Bob, I never saw it again.

I hooked up the test boxes to give me an LC filter into the woofer and a CLCRR into the tweeter (**Fig. 1**). I turned on the dastardly black box and adjusted the frequency to a 2Hz pulse rate and listened. At first I thought I had made a wiring mistake because all I heard was bass. After disconnecting the woofer, I traced out the tweeter wiring and found I had set the “Waveform” to low-frequency pulse.

I set it to the 252μS exponential pulse and then adjusted the tweeters’ crossover components for the cleanest click I could get. I then hooked the woofer back up and changed to the 830μS weighting on the pulse. After setting the capacitor in the woofer circuit to the middle of its range, I closed my eyes and adjusted the woofer inductor for the cleanest click. Then I tweaked the capacitor.



THE FINAL TEST

I then set up my field test kit—an Earthworks M30 microphone and an M-Audio USB preamp to my Dell laptop running Smaart. For the signal source I used a Sony SCD595 SACD player, my custom built Holco resistor attenuator, and my Class A amp. The curve was within ±3dB from 150Hz to 20kHz. The limitations were the driver, box, and location!

With my field kit I could calculate the adjustments to the crossover to get even better response and then try them out. But the Sound Strobe demonstrated that even technically blind it would allow

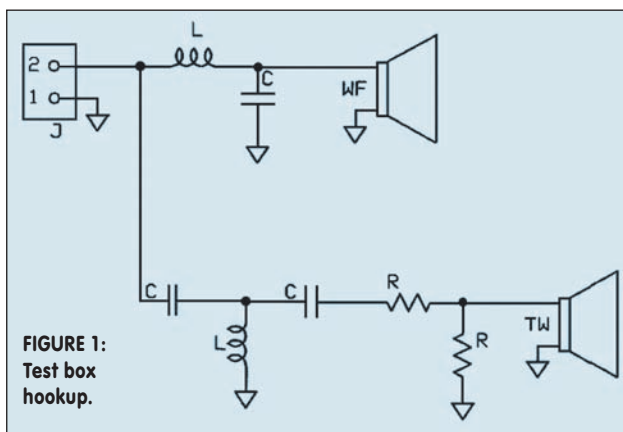


FIGURE 1:
Test box
hookup.

me to get the best possible settings with just one box! Playing with the crossover topology and values, the Sound Strobe would take just a bit longer.

The question that remains is, can you use music to make these same adjustments? One of the concepts that is currently exploited in audio measurement systems is to use processing power to compare a reasonably simple signal such as pink noise and to the loudspeakers’ output and compute what the difference is.

The other school of thought is to make

the test signal complex and the measurement gear simple. Dennis Colin has not just done the second method, but he has nailed the variations in waveform to maximize the results. You could use music, but it would take much longer.

The Sound Strobe can do even more! I am not going to reveal all its capabilities. It takes two days for me or my competitors to tune up a big sound system, and when we are done we

really have tuned only half of the things that should be adjusted. This unit will let me tune the other half and I am not about to disclose in print what my competition does not know. You will have to find out for yourself! *ax*

The Sound Strobe is available in kit or assembled units from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, 888-924-9465, e-mail: custserv@audioXpress.com.



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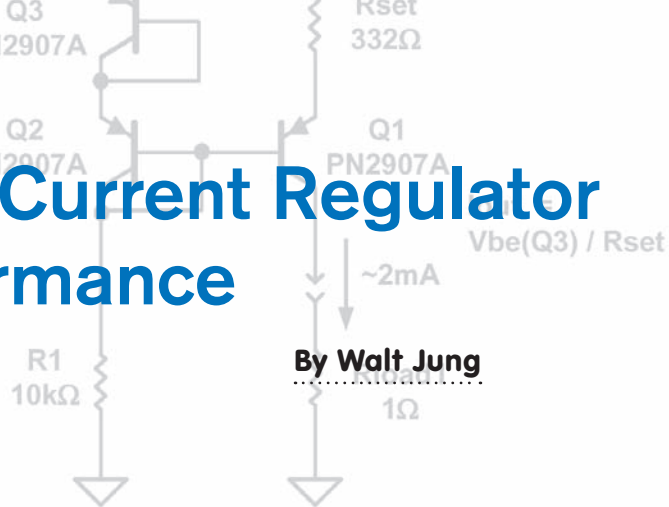
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Sources 101: Audio Current Regulator Tests for High Performance

Part 1: Basics of Operation

Noted designer and author of a classic op-amp design cookbook begins his series on quieter power supplies.



By Walt Jung

This article deals with audio current regulator circuits and their performance with regard to power-supply-related noise. It describes a test methodology to characterize these audio current regulation elements and circuits for sensitivity to applied voltage. In terms of audio power supply circuitry within systems, this would be the *line rejection* or *impedance properties*. Because audio current regulators are often referred to as *current sources*, and the performance aspects of such for audio are basic, the article is logically entitled “Sources 101.”

But, it isn’t the last word on the topic by any means. I hope it can explain to many how to build better current regulator circuits for audio. And, most important, it definitely shows how to test these circuits, and to differentiate their performance.

THE WHYS AND WHEREFORES

While I have been fascinated by current regulators for years, it wasn’t until recently that I investigated deeply the wideband AC performance of various topologies operating at many current levels. It was brought home to me that some greater insight could be useful, when a relatively simple “One V_{be} ” type of current regulator was found lacking, as operated within a shunt regulator design. That experience started this investigation several months ago, and the article here is one fruit of those explorations.

So, there we are. It may seem obvious that it is desirable to have audio circuits with high immunity to power supply potentials, but this area is seldom, if ever, addressed in audio design and construction articles. Typically, a current

source or sink circuit is presented without related performance data, so it is difficult to differentiate what types of designs are better for audio. Some relatively simple power-supply rejection tests can help differentiate which circuits perform best and are most immune to power-supply-related distortions. This article details a wide variety of available current regulation circuitry, and documents their relative performance in terms of power supply immunity.

CURRENT REGULATOR PROPERTIES FOR AUDIO

Current regulators have several operating parameters that are key to audio circuit performance. Probably the most important is the *degree of regulation* of the associated current. In other words, the output current, I_{out} , is maintained at some fixed design level, one which is relatively independent of other circuit conditions. Thus, although the unregulated input voltage varies, the design current stays constant.

As noted, just how well this is done is one of the major figures of merit for a current regulator. You can detail the specification for this in several ways. One is simply the rejection (or line rejection) of noise appearing on the input, expressed most simply in dB, such as, for example, a rejection of 100dB. This means that input noise is reduced by 100dB ($100,000/1$), or it becomes -100dB with respect to the input ($1/100,000$).

Because audio is a wideband signal, it is important to quantify a regulator over at least the 20Hz–20kHz audio bandwidth, and preferably even wider. This is because these circuits often have

a nonlinearity that can be excited with supersonic signals. This particular performance parameter is also expressed as *impedance*, which I will discuss further.

One form of nonlinearity often found in solid-state audio circuits is nonlinear capacitance. While a pure capacitance is not a distortion producer within an audio circuit, nonlinear capacitance can—and will—produce distortion, particularly when excited with high frequency (HF) noises, such as those typical to rectified-AC supply systems. So, one indirect figure of merit for audio current regulators is the associated capacitance. The lower this is, generally the lower will be any spurious responses to HF noise.

CURRENT SOURCE OR CURRENT SINK?

One point of potential confusion regarding current regulators for use within audio circuits lies with the terminology used. Two terms you often see are *current source* and *current sink*. These terms are often used interchangeably, but this isn’t always technically correct. Some review of the terminology is helpful.

A *current source* circuit is a device (or more complex circuit) that provides current regulation properties, typically operating between a positive rail voltage and common (ground), often using one or more PNP transistors, so as to *source* load current. This is the more popular usage. However, you should note that current regulator circuits operating between a negative rail and common are also sometimes called current “sources.”

Regardless of this definition muddling, it is more accurate to describe such circuits as *current sinks*; that is,

a current regulator biased with respect to the negative rail, often using one or more NPN transistors to *sink* load current. This is the terminology I will use in this article; i.e., a *current source* operates between a positive rail and the load or common, and a *current sink* operates between a negative rail and the load or common.

To add slightly to the confusion, there are single transistor devices that, because of their unique bias flexibility, can operate as either a current source or sink. An example would be the *common junction field effect transistor*, or JFET. So, in applying such devices, their explicit connection details will determine their exact function. More specifically, N-channel JFET devices such as the J202 and others exhibit this type of flexibility, and will be illustrated shortly.

WHAT TESTS?

While there are many tests potentially useful for audio characterization, this exercise concentrates on power-supply rejection versus frequency, which yields a picture of current regulator impedance versus frequency. These two performances go hand-in-hand. For best immunity to power rail noise components, a wide rejection bandwidth is desirable within the audio circuits. This may not be immediately apparent, because unregulated audio rails usually have predominant 120 or 100Hz ripple, from which it is relatively easy to provide immunity.

A salient point here is that the AC to DC rectification process is by definition a wideband noise generator, by chopping the AC mains waveform into high peak current pulses in the typically used capacitor input filter. A Fourier analysis of the noise components will show there are ample HF components associated with such supply rails, not just the 100/120Hz fundamentals.

Another subtle point is that the audio circuits themselves don't have infinite bandwidth. Thus, while they may have some degree of good supply rejection at lower audio frequencies, in the ultrasonic range this rejection deteriorates, introducing the potential for power noise components to *intermodulate* with the audio. Of course, for high quality reproduction, the possibility of any such intermod must be minimized, either by careful filtering or the development of circuits intrinsically immune to the nonlinearities that can produce the intermodulation.

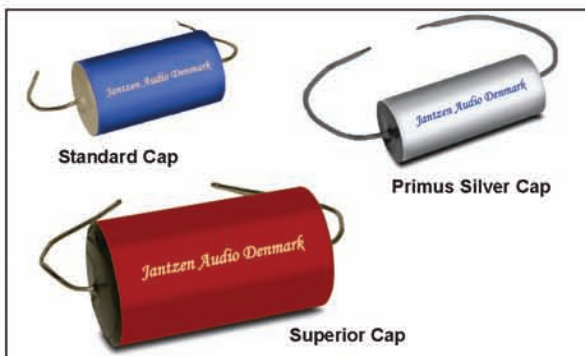
TEST SETUP

The general setup for this test series is as per Figure 2a of Reference 1, available at www.waltjung.org/PDFs/Regs_for_High_Perf_Audio_1.pdf. The test is basically a highly sensitive crosstalk measurement done in the analog domain, measuring the output of a circuit as it is related to a 20Hz-200kHz swept sine wave excitation. This method was originally developed for the 1995 series of articles on audio regulators¹.

In this current series of tests, the "regulator" under test is either a current source or sink, which is connected as per **Fig. 1**. Here, the device or circuit under test is a two-terminal (in some cases, three) current source device (or circuit) connected between Vin and the load. The load is simply a 1Ω resistor, Rload1. Note that unlike Fig. 2a of Ref. 1, no output load

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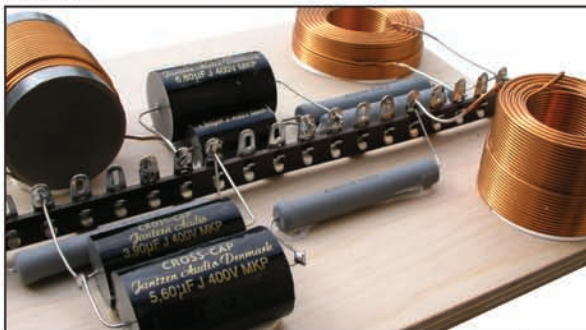


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capacitor is used for this test. The rail driver circuit supplying the +18V DC excitation with a superimposed 1V RMS AC swept frequency signal is shown in Fig. 6 of Ref. 1. It is represented here as the dashed box figure.

Calibration of this test setup is done by first calibrating the Audio Precision test set to the 1V RMS 0dB level, against the applied rail driver $V_{in}(AC)$ signal. All subsequent measurements are then referenced to this 0dB level. The calibration is completed with a series of test runs, using fixed 1% calibration resistors in the Device Under Test (DUT) position. This step establishes corresponding reference V_{out} levels of 10k Ω , 100k Ω , and 1M Ω , as shown in the like-named plots of **Fig. 2**. On the dB scale, these correspond to levels of -80dB, -100dB, and -120dB, respectively.

In this figure there is also shown a much lower level trace, marked GND. This trace is the lower limit of the test setup dynamic range, which is the residual noise seen at V_{out} of **Fig. 1** with the test set active and no DUT connected. In essence, virtually all practical devices/circuits tested show appreciably higher levels at V_{out} , although some do indeed begin to approach the full dynamic range of more than 140dB.

As you can also note in **Fig. 2**, the plot for the highest fixed calibration impedance of 1M Ω falls about 20dB (or more) higher than the residual noise shown in the GND trace, at the lowest frequencies. At the higher frequencies, setup/system noise shows impedance lowering, corresponding to a dynamic range of ~110dB at 200kHz, which is roughly equivalent to 300k Ω .

THE MEASURED NOISE

The nature of this test is to measure, first of all, *synchronous noise*. This is because it is a modified crosstalk test, and operates by sweeping a

measurement bandpass filter, looking at the spectrum appearing across R_{load1} . The measured noise, however, can be from either of two sources. The obvious one is due to the synchronous components related to V_{in} ; i.e., the “crosstalk” noise components¹ (see “Hofer” box, p. 8).

But, another potential component could be due to any *self-generated* noise in the DUT. The test as configured here really has no means to distinguish one noise from the other—they are simply lumped together. It is true, however, that if and when a very low noise level is measured, both the synchronous as well as the self-generated noises must be low. And, in the many tests that follow, many circuits show very low noise—meaning that they have at or approaching levels of -140dB with respect to 1V RMS; in other words, on the order of 100nV RMS. A potential future investigation might examine self-generated noise

more closely, with a higher load impedance and noise analysis software.

WHICH DO YOU PREFER: IMPEDANCE OR NOISE REJECTION?

You should understand that current regulator circuits may be specified in terms of either equivalent dynamic impedance; i.e., “100k Ω ,” or rejection with respect to some applied voltage reference level. Here, the three calibration impedances equate to rejections of 80dB (10k Ω), 100dB (100k Ω), and 120dB (1M Ω).

For impedances (Z) of more than 10k Ω (rejection of more than 80dB), you can use one of these approximations to convert between the two terminologies.

$$\text{dB} \sim 20 * \log(1/Z)$$

or

$$Z \sim 10^{(\text{dB}/20)}$$

In the test plots following, the current source or sink can be either a simple device such as a JFET, an IC such as a three-terminal regulator operated as a current source, or a more complex circuit comprised of transistors, resistors, and so on.

Referring to **Fig. 1** again, note that the measurement point of R_{load1} is fixed with respect to the common point. For current source circuits, this is most appropriate, and the output current is measured directly and accurately. For some current *sink* circuits, a more appropriate current sample point would appear to be with R_{load} to the (+) lead of the DUT. This test setup doesn't provide for this, but nevertheless current sink type circuits can still be assessed. In the actual measurements that follow, this detail will become somewhat moot.

Referring to **Fig. 1** again, note that the measurement point of R_{load1} is fixed with respect to the common point. For current source circuits, this is most appropriate, and the output current is measured directly and accurately. For some current *sink* circuits, a more appropriate current sample point would appear to be with R_{load} to the (+) lead of the DUT. This test setup doesn't provide for this, but nevertheless current sink type circuits can still be assessed. In the actual measurements that follow, this detail will become somewhat moot.

THE MEASUREMENTS

I measured a wide variety of current regulator circuits, all of which were built up on 8 pin header assemblies and plugged into the **Fig. 1** setup as a DUT. In the performance plots which follow, the

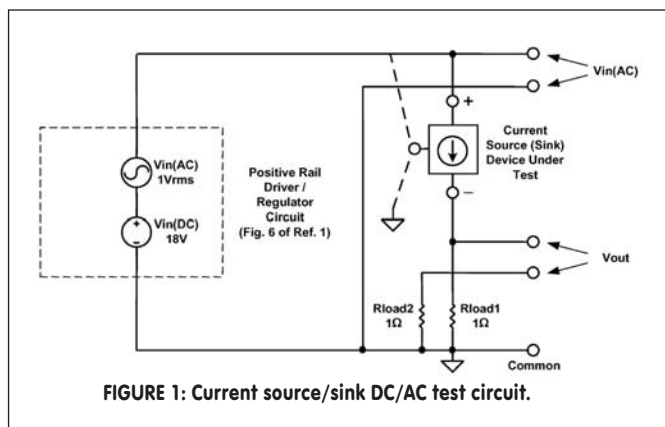


FIGURE 1: Current source/sink DC/AC test circuit.

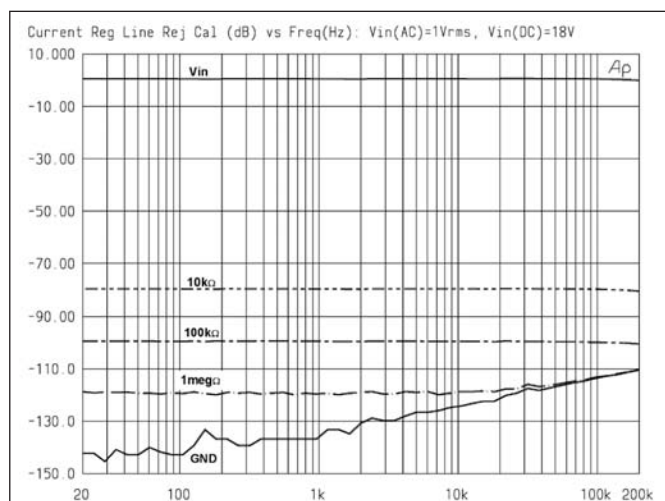


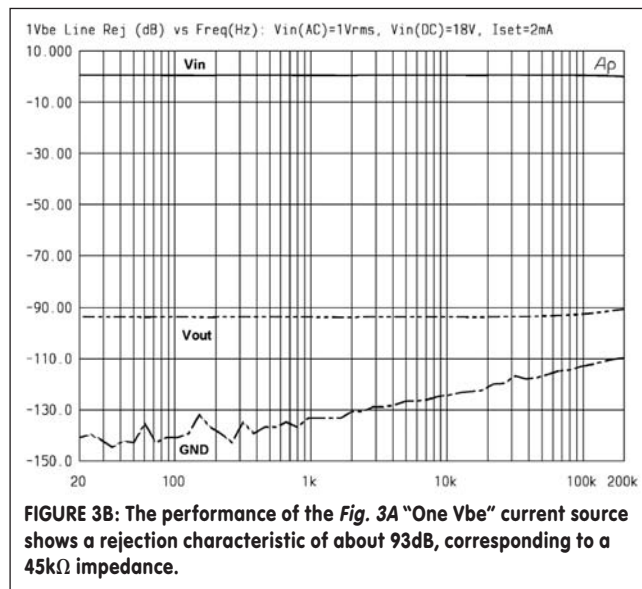
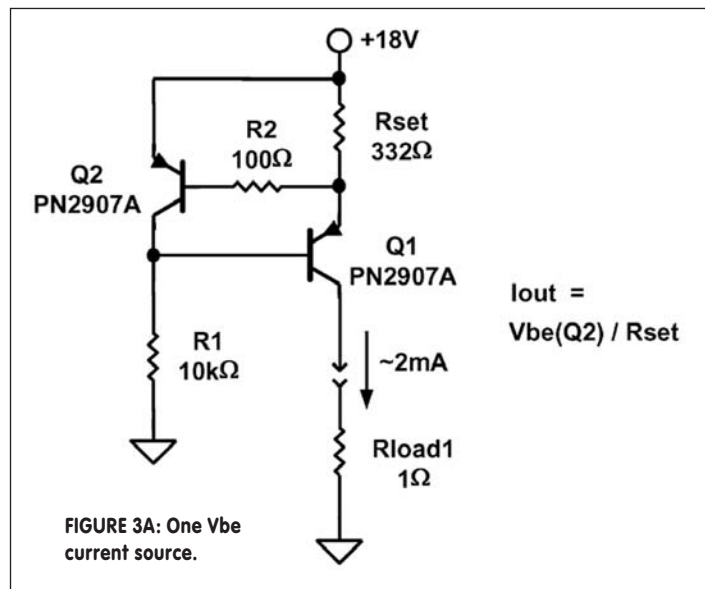
FIGURE 2: Multiple calibration runs with fixed 1% 10k Ω , 100k Ω , and 1M Ω resistors establish measurement validity. The dynamic range limits correspond to V_{in} and GND.

Vin(AC) trace is shown at the top, and the GND trace at the bottom. The actual measurement data for a given DUT is contained in one (or more) Vout traces, which fall somewhere between the upper/lower dynamic range extremes. For cases of multiple Vout traces representing different conditions, each is labeled for clarity.

One Vbe Current Source

A "One Vbe" current source is shown in Fig. 3A, based on PNP transistors. In this and many circuits that follow, the transistors used are the high gain, TO-92 versions of general-purpose industry standards, either the 2907A series for PNPs, or 2222A series for NPNs. For general purpose work, these parts are

preferable, since they represent a sweet spot of performance for these applications, that is excellent linearity for good rejection at lower frequencies, yet still low-to-moderate capacitance, which allows the good performance to hold up well with increasing frequency, at great prices! They allow currents from low μA levels up to 20mA or more, at voltage



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levels of 40V (or more).

Of course, higher-voltage parts should be used when appropriate. While exotic and super-high-gain parts aren't necessary for very good performance from these circuits, low capacitance devices definitely are preferred (<10pF), a critical point if substituting. For truly excellent performance (at higher cost and reduced availability), you can use select "2S" series parts. Notable examples here are the Sanyo 2SA1016 PNP and 2SC2362 NPN (see www.semiconductor-sanyo.com/discrete/index.htm). These offer lower capacitance than the 2907/2222 families, and are useful up to 120V or more. They can work in the circuits shown, substituting for the PN2907A and PN2222A, respectively.

The familiar circuit of **Fig. 3A** is often

used in audio circuits, perhaps due to the relative simplicity. It is configurable over a wide range of output current levels by adjusting the sensing resistor, R_{set} , which drops one V_{be} (that of Q_2) in operation. As shown, the output current is about 2mA.

The PN2907A types are useful up to medium voltages and several tens of mA. Q_1 may require a heatsink for power dissipation of 0.5W or more. You can make a mirror-image current sink circuit with NPN transistors such as PN2222As for Q_1 - Q_2 , referring R_{set} to a negative supply. Note: This circuit can oscillate readily when using wideband transistors. The tendency to do so is suppressed by R_2 , which should be used with any form.

Performance of the **Fig. 3A** current

source is shown in **Fig. 3B**, a plot of V_{out} for the cited test conditions. While the impedance offered by this circuit is rather independent of frequency, being flat to nearly 200kHz, it really isn't all that high. The rejection of ~93dB corresponds to an impedance of about 45k Ω .

While at first glance this might appear OK, the performance of this circuit is easily bettered by many others, both more simple and cheaper. For these reasons, plus the stability caveat, this circuit isn't recommended. Except as an example to avoid, perhaps!

Two Diode Current Source

An effective and still relatively simple current source is shown in **Fig. 4A**, again based on PNP transistors. With Q_2 and Q_3 connected as diodes, this is a

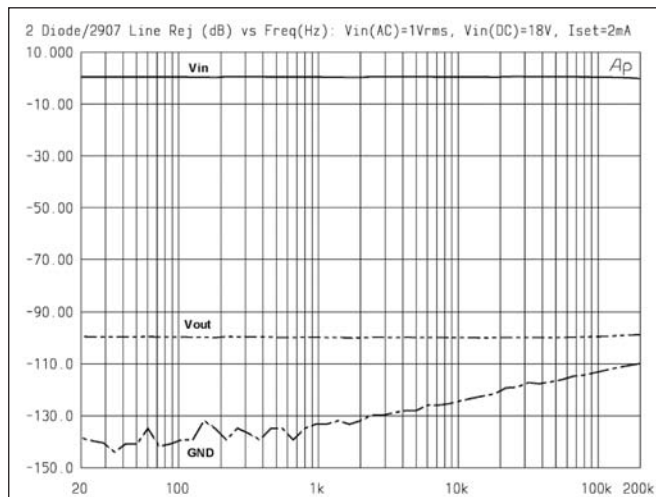
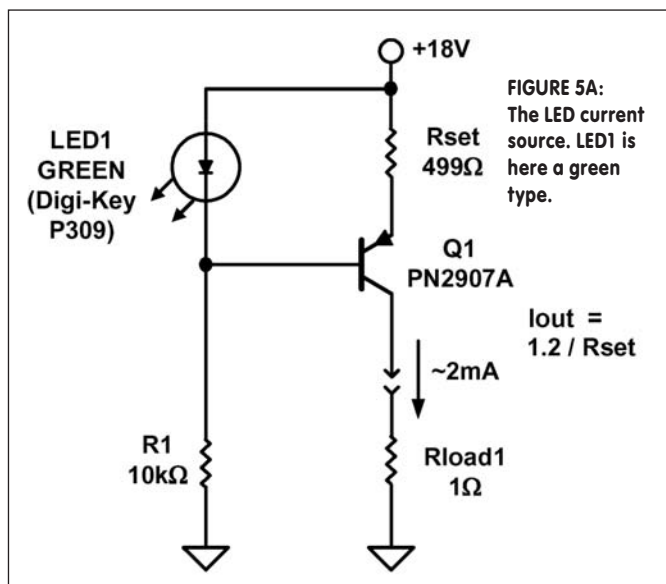
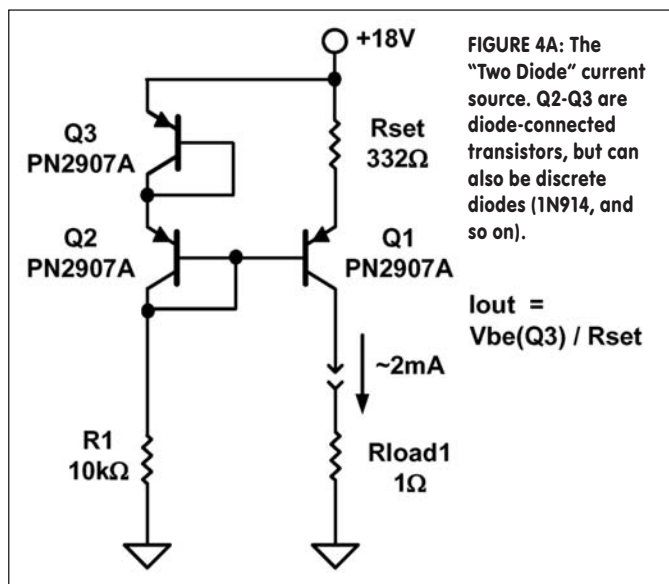


FIGURE 4B: The performance of the **Fig. 4A** Two-Diode current source shows a rejection characteristic of about 100dB, corresponding to a 100k Ω impedance.

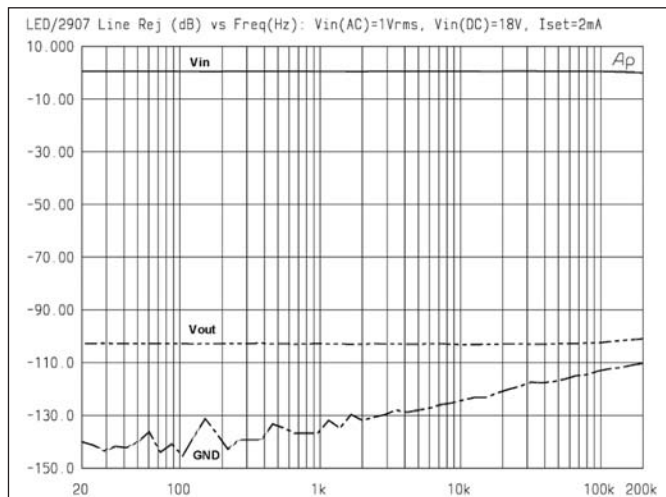


FIGURE 5B: The performance of the **Fig. 5A** LED based current source shows a rejection characteristic of 105dB, corresponding to a 177k Ω impedance.

two-diode-biased current source. It is often seen with two 1N914 or other diodes functioning just as Q2-Q3 do here. While it does use three transistors, it can still be inexpensive, because standard types such as these PN2907As are less expensive than metal film resistors. Output current is set by R_{set} and the V_{be} of Q3, and is 2mA. As was true with the One V_{be} current source, you can implement a current sink with the use of PN2222A NPN transistors.

Performance of the Fig. 4A current source is shown in Fig. 4B, where the impedance is again independent of frequency. Here the rejection of 100dB corresponds to an impedance of 100k Ω . This circuit is useful for moderate performance at low cost, at the expense of requiring five parts. Or, you could maximize efficiency by using the dual (or quad) packaged 2907A transistor types.

LED Current Sources

Replacing the two diodes of Fig. 4A with a single green LED forms the LED current source of Fig. 5A. This setup is slightly less complex and inexpensive,

and provides good performance for the cost/complexity. Output current is set by the value of R_{set} , which has about 1.2V across it. The slightly higher voltage of a green versus red LED provides the 1.2V, but you could also use a red LED. A current sink is also possible, using NPN transistors biased to a negative voltage, plus, of course, an appropriate polarity change for the LED.

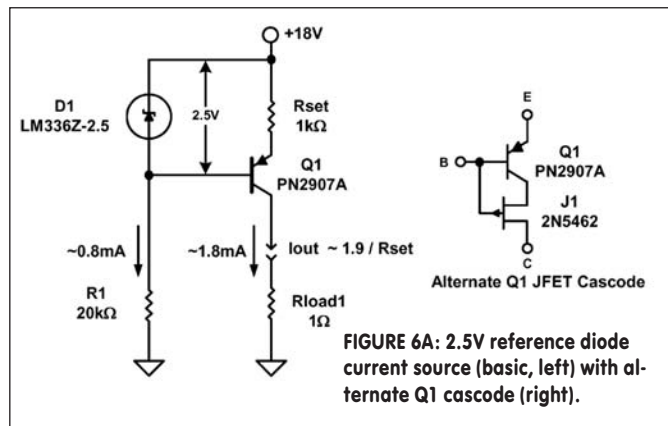
Performance of this LED current source is shown in Fig. 5B, operating with an output current of 2mA. The rejection is about 105dB, corresponding to an impedance of 177k Ω . This circuit actually achieves better performance than the "two-diode" current source, but with fewer parts!

Reference Diode Current Source

If you replace the LED (with the relatively poor ref-

erence voltage) from Fig. 5A with a true voltage reference diode, a much higher quality current source is formed (Fig. 6A). Note that this form of the circuit has the same number of basic parts as the simple LED source, but is capable of much higher performance. This is due largely to the more stable voltage across the diode, which changes very little with current. An optional cascode connection for Q1 enhances performance considerably.

In this example an LM336Z-2.5V diode is used, resulting in about 1.8-1.9V



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across R_{set} . Thus a $1k\Omega$ value for R_{set} supplies just under $2mA$ of output current. Note that the circuit is by no means limited to just such lower currents. As long as the power dissipation of Q_1 is maintained low (or sufficient heatsinking used), currents up to $10mA$ or more are allowable from this basic circuit.

Figure 6B shows the performance of this simple current source in two modes, one basic, the other with Q_1 operated with the optional cascode. In both cases the resulting performance is excellent. For the basic operation using a simply connected PN2907A for Q_1 as shown at the left, the low frequency (LF) rejection approaches $130dB$, which would be equivalent to $3M\Omega$. There is a slight rise in impedance at the upper frequencies, but all in all, the performance is exceptional for such a simple circuit.

When the optional cascode connection shown at the right is used, both LF and HF performance is enhanced. The cascode data, as noted, approaches the residual noise at all frequencies. It is enabled simply by using the optional connection for Q_1/J_1 , which is, in turn, used as Q_1 in the main circuit.

Note that this rather elegant connection is *self-biasing*, due to the manner in which the V_{gs} of J_1 automatically provides a collector voltage for Q_1 . It will do so as long as the I_{dss} of J_1 is equal to or greater than the desired output current. For example, the 2N5462 specification for I_{dss} is $4mA(min)$, which means that this cascode should only be used with lower currents, such as this $1.8mA$ case.

Alternately, for higher currents, either 2N5462s can be screened for a higher I_{dss} , so that the tested I_{dss} of J_1 is always equal to or more than the desired I_{out} . Or, a basically higher I_{dss} family of P-channel FETs can be used, such as the 2SJ74 V series (being careful to note that these are limited to $25V$ supplies). But, the 2N5460/5462 series is preferred, both from the standpoint of the low capacitance they offer, $<10pF$, plus their $40V$ voltage rating. The low noise 2SJ74 audio types have much higher capacitance than $10pF$ and don't perform as well as HF current sources (although they do excel below $1kHz$).

With a cascode connection such as this, you can expect a loss of output swing, due to the gate-source bias voltage of J_1 . Take

care in testing at higher output swings to ensure that this is not a problem. The circuit of **Fig. 9A** following also discusses cascodes, at higher output currents.

Although the D1 reference diode used in **Fig. 6A** has low TC by itself, as it is applied here the net output current will still change with temperature, following a positive slope due to the V_{be} of Q_1 . Another version of this circuit can also be built, using a $1.2V$ diode for D1. With this option, the output current will also have a positive slope of about $0.3\%/^{\circ}C$; i.e., the opposite of that of a conventional silicon diode. Although none of the current sources/sinks discussed thus far have low inherent TC, several examples to follow do in fact have both low TC and a predictable output current.

LM334 Current Sources and Sinks

The LM334 is a monolithic IC designed to be used as a current source (or sink), or, alternately, as a temperature transducer. It is shown in **Fig. 7A** (left), used as a basic $2mA$ current source, requiring only one additional part for functionality, R_{set} . The device features very simple operation and can be used up to about $5mA$ of current as shown. It is perhaps one of the more predictable types for output current among the circuits described thus far.

But, note that the LM334 output current is *not* constant with temperature; it does, in fact, vary linearly with a positive slope of about $0.33\%/^{\circ}K$. The basic expression for output current is shown in the figure. Note that the very low sense voltage implies efficient utilization of power supply voltage. The part achieves operation at low thresholds, requiring only $\sim 1V$ across the terminals to operate. Just as shown here, it is configured as a current source. Operation as a current sink is possible with the same number of parts, but with the V- terminal and R_{set} tied to a negative supply, with a load connected to V+.

Rejection versus frequency performance of the LM334 as both a basic and a cascode current source is shown in **Fig. 7B**. As will be appreciated, these impedance characteristics are exceptional, particularly at the lower frequencies where rejection is a few dB above the setup residual noise, even for the basic curve. This is indicative of an equivalent

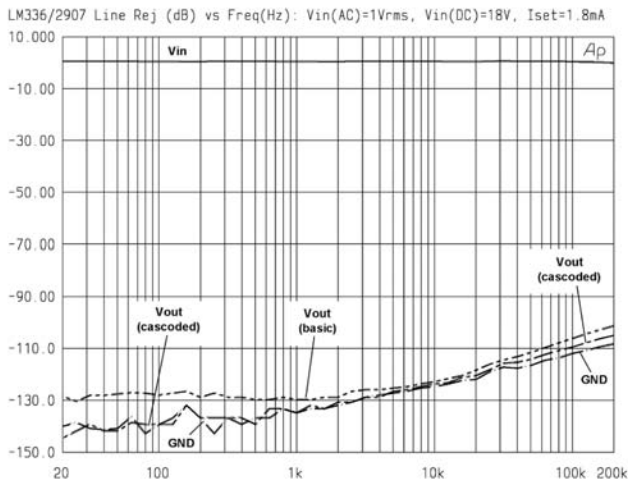


FIGURE 6B: The Fig. 6A reference diode current source performance for both basic and cascode operating modes. Cascoding greatly improves low frequency rejection.

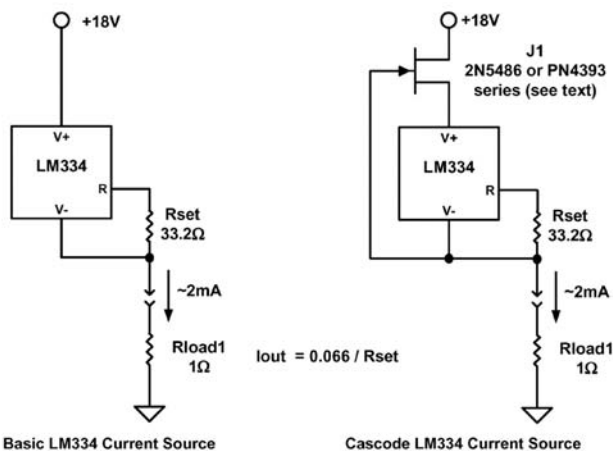


FIGURE 7A: LM334 current source (basic, left) with alternate cascode (right).

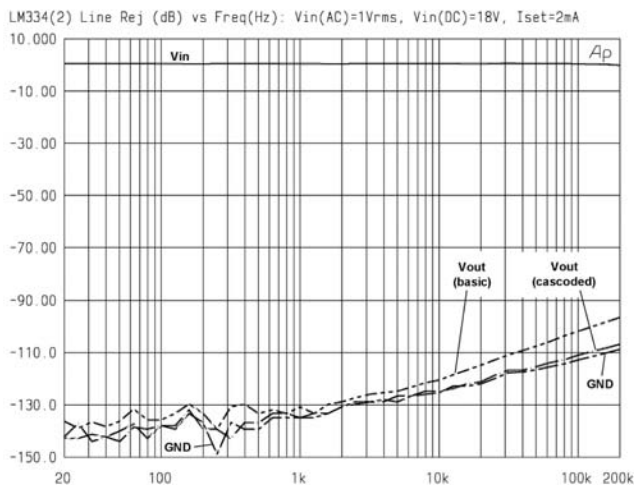


FIGURE 7B: The Fig. 7A LM334 current source performance for basic and cascode operating modes. Cascoding improves both low and high frequency rejection.

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impedance close to 10M Ω , albeit with deterioration at the higher frequencies.

An improved version is available by adding a cascode-connected JFET, shown as the **Fig. 7A** option at the right. With this version, J1 is preferably a 2N5486 JFET, which is suitable for output currents up to 2mA. There are no other changes to the circuit. The beauty of this cascode connection is that it extends both low and HF impedance, and the resulting composite performance is barely above the residual noise level. The circuit can still be used either as the current source shown or as a current sink, with the load in the drain lead of the JFET.

However, it is important to note that selecting the JFET device for the cascode position really isn't trivial, but requires some careful thought. The 2N5486 series is quite useful because of the low capacitance and medium currents, *but it does have the disadvantage of only a 25V voltage rating*, lower than the LM334's 40V rating. Alternatively, the PN4393 family can operate to 40V, but it does have higher capacitance than the 2N5486 family of RF devices.

Other JFET parts can also work as J2. The basic requirement is that the desired output current be safely less than the minimum device Idss rating, and the JFET Vgs at the operating current is safely greater than the LM334 minimum voltage of 1.2V (~2V recommended). Some general considerations for the JFET selection are included on the LM334 datasheet², and in references 3 and 4. This process

is also discussed in more detail with the JFET-based current sources to follow.

Some LM334 caveats: As typically operated, an LM334 produces noise components higher than that of a simple bipolar transistor at the same current. This is fundamental to the device design, so take this factor into account before application in low level circuits (see datasheet discussions). It will also have a higher output capacitance than many small signal transistors—about 15pF. Fortunately, this latter characteristic can be mitigated with the use of cascoding, providing that the cascode transistor used has low capacitance.

JFET-Based Current Sources and Sinks

It is fairly well known that many JFET devices make natural current limiters, and thus they can be used as either current sources or sinks. For an N-channel part (the most popular) all you need to do is short the gate and source terminals, apply positive bias to the drain terminal, and connect the gate/source to the load. Under these conditions the JFET will conduct a current equal to the device's Idss (drain current with gate/source common). Once the applied voltage is greater than a minimum voltage equal to Vgs(off), this current will then remain relatively constant with further voltage increases; i.e., the device is operating in a *current-limited mode*.

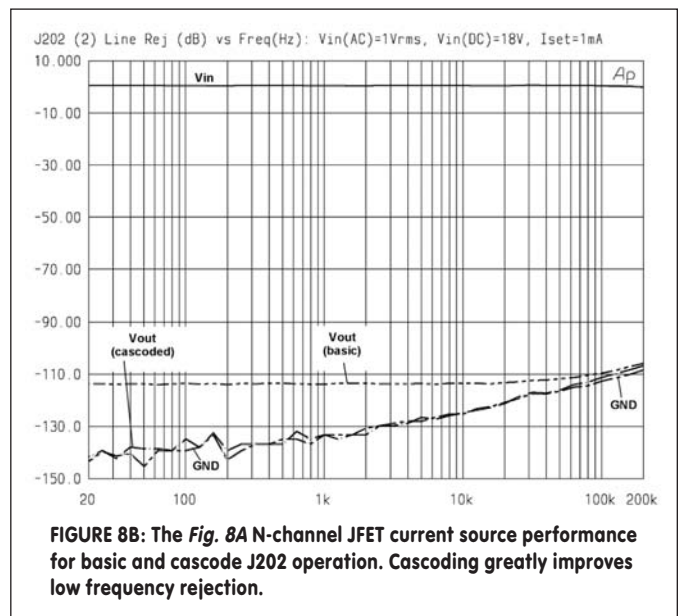
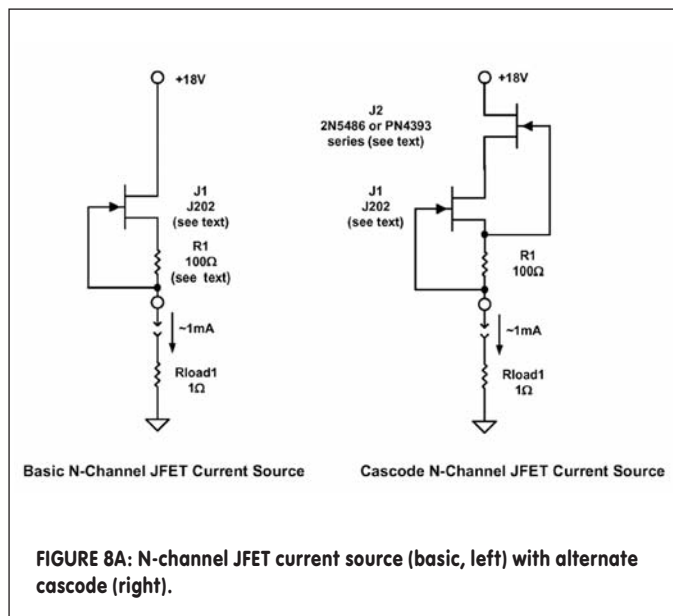
What is not likely to be as well known are the details appropriate to selecting a JFET part so that this supply voltage independence is maximized. For the

testing described here, this implies the highest rejection characteristic. This is a function of both the specific JFET part itself, and how it is used.

A case in point is the simple JFET current source of **Fig. 8A**, using the J202 N-channel FET. While many different FETs can be used in this circuit, the J202 series is well suited to this application, because the rejection characteristics are optimum for the J201 and J202, which are 40V rated devices. They come from a family made with a *long gate* process, which minimizes the device output conductance, and thus maximizes the rejection. *An important note: References 3 and 4 cover this area quite well, and should be considered required reading for anyone building these types of circuits.*

Within all the various JFET process families, there are typically several classifications of devices, sorted as to Idss and Vgs(off) limits. The lower Vgs(off) parts in a given family will have the lowest output conductance, and thus the best rejection. For this family, this would be the J201, which has a max Vgs(off) of 1.5V. From Reference 3, for best rejection, the J1 Vdg > Vgs(off), preferably 2 * Vgs(off). Practically speaking, this means that such FET circuits like lots of voltage to achieve their best performance.

But, it isn't always possible to use such a low Vgs(off) part, as in this case I needed 1mA, which is well above the J201's minimum Idss of 0.2mA. So I used the J202, which, fortunately, still gave great results.



Although the most simple circuit of this type would use an R1 value of zero, I set up these tests with $R1 = 100\Omega$ to simplify selection of a device sample to conduct 1mA, the target output current, which occurs with 0.1V across R1. This device was then used in the rejection tests. It actually helps the circuit a bit to use some finite resistance for R1, because this resistance increases the net output impedance slightly. Consider it optional in your final circuit. Store-bought JFET current limiters simply short the gate-source terminals and sell the so-wired part as a two-terminal device (equivalent to the J1 part of the **Fig. 8A** circuit with R1 shorted).

Performance of the J202 as both a basic and cascode JFET current source is shown in **Fig. 8B**, and as you can note, even the basic circuit is excellent, considering the simplicity. The rejection is about 113dB, corresponding to roughly a 446k Ω impedance. While this can be considered very good for such a simple circuit, this is really due to the optimum conditions. There is first the optimum type of device (the J202), then there is also the relatively high bias voltage of 18V, well above the 2 * $V_{gs(off)}$ rule of thumb. Circuits which

bias J1 at lower potentials would be expected to perform more poorly, particularly when V_{dg} approaches $V_{gs(off)}$.

While the J202 current source in basic mode is excellent for the simplicity, adding a cascode device makes the rejection characteristics approach the measurement limits. A cascode version of this 1mA current source is shown within the right option of **Fig. 8A**. These measurements used a 2N5486 for J2, and the cascode mode data of **Fig. 8B** is barely discernible from the residual noise.

But, there are more caveats to remember about this cascode circuit. The cascode device must be selected to provide a bias for J1 that is above J1's $V_{gs(off)}$, preferably well above. For the sample devices used here J2 applied a 2.7V bias to J1, indicating that the sample J202 was a low $V_{gs(off)}$ device. Furthermore, the 2 * $V_{gs(off)}$ rule should also be applied to J2, and will be more stringent for that part, because it is by necessity a higher $V_{gs(off)}$ part than is J1. The comments about deterioration of rejection with lower rail voltages apply even more so to this cascoded version, and may require supplies of 12V minimum, for example, for the very best

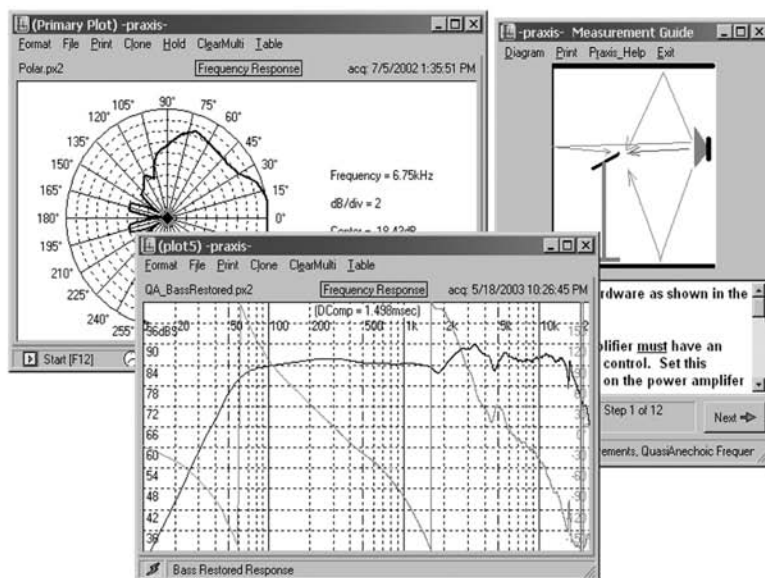
performance.

So, in essence what you have here is a circuit that can work really well, but also exhibits some subtlety to extracting maximum performance. In fact, some J1/J2 pre-selection may be appropriate to set things up optimally. Fortunately, it appears that once the J1/J2 DC biasing conditions are properly met, the good AC performance falls in place. Watch the voltages on J1 and J2 and use higher voltage parts when necessary for the cascode. A caveat here applies to the 25V 2N5486, as noted.

Finally, I reiterate that both the basic and the cascoded versions of this current regulator can operate as either a source (as shown) or as a sink, with the JFET most negative terminal tied to the negative supply and the load in the positive leg. This is one intrinsic beauty of all JFET-based current regulators; that is, the ability to operate as either a source or a sink, without performance compromise. Note a somewhat subtle minor point here: you need only consider N-channel JFETs for such two-terminal current regulators; not only do they work (well), there are many more of them to choose from than the P-channel counterparts.

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JFET Current Regulator Diode Based Current Sources and Sinks

Most of these discussions apply equally to JFET *current regulator diodes* (also called JFET current limiters), because they are JFETs internally wired as two-terminal parts. One series of these is the Vishay/Siliconix J500 family, consisting of individual parts with output current ratings from 0.24mA (J500) to 4.7mA (J511), all rated for 50V operation⁵. Of this series I had some J507s on hand, rated for a nominal 1.8mA operation, and some of these were tested.

Applying this type of current regulator is simplicity itself, as shown in the simple two-part basic test circuit of **Fig. 9A**, using a J507 as J1. Virtually all of the performance characteristics of this circuit

are dependent upon the basic J1 part (and to some extent the applied voltage). In the case of the J507 sample, an output current of ~1.5mA resulted.

Figure 9B shows the performance of this circuit for both basic and cascode modes. For the basic mode, the LF rejection is 120dB, equivalent to a 1MΩ impedance. Even at the higher frequencies the impedance is still excellent, rising only slightly above the noise level.

The two-terminal J507 is cascoded with a JFET as shown as the right option in **Fig. 9A**, a hookup very similar to the JFET cascode of **Fig. 8C**. For the J500 series, a minimum limiting voltage bias is given on the datasheet, which for the J507 is 2.5V. Thus any cascode circuit should provide a bias from the cascode part above this voltage to achieve best results.

The actual voltage for the parts tested was 2.6V, which was sufficient to allow the performance shown in the cascoded Vout plot. As you will note, this allows a LF rejection of ~140dB (10MΩ), and at higher frequencies performance just above the noise level.

Like the cascoded JFET circuit of **Fig. 8A**, you can apply this cascode to any part of the J500 series or to other similar JFET current regulators. You should choose the cascode device to supply the required minimum voltage across J1. Also take into account voltage limitations for J2, as in the case of the J2 device of **Fig. 8A**, right.

Reference Diode Buffered

Current Source

The necessity for very careful attention with the JFET cascode biasing can be a

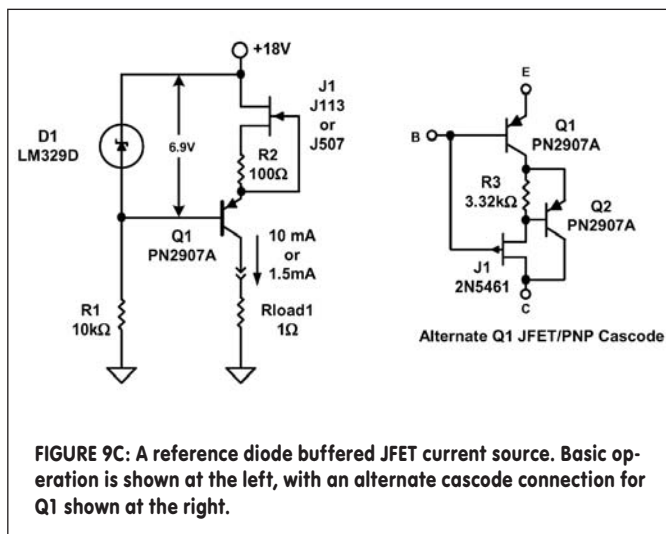
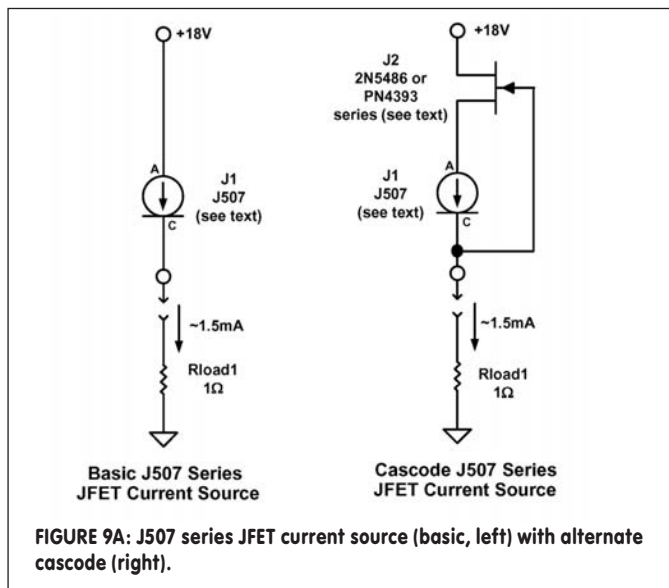
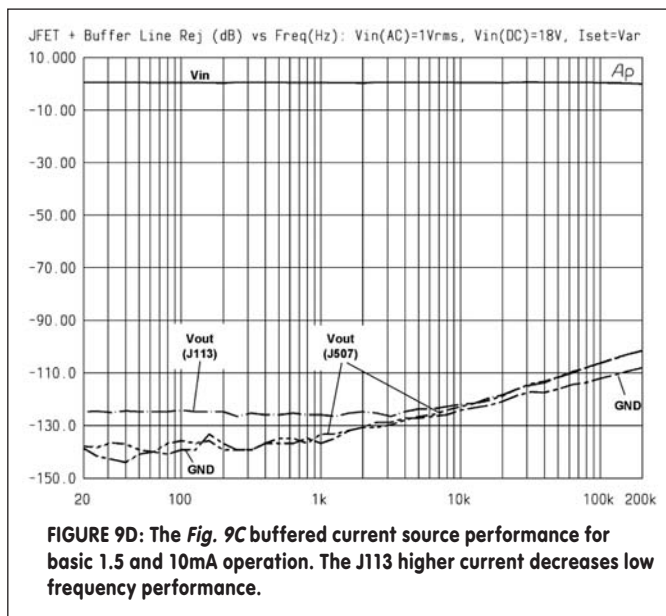
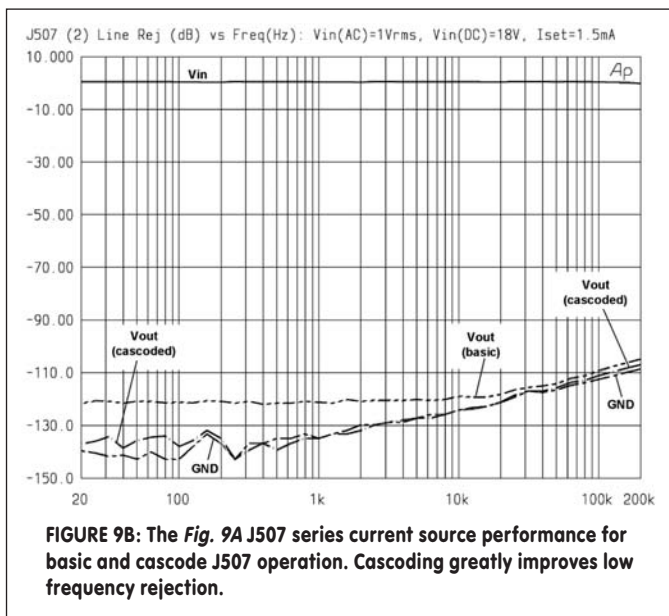


FIGURE 9C: A reference diode buffered JFET current source. Basic operation is shown at the left, with an alternate cascode connection for Q1 shown at the right.



real source of user frustration. Or, perhaps, a much higher voltage capability may also be necessary to get around a low JFET device rating, for example. A solution to both of these points is the reference diode buffered current source of **Fig. 9C**.

Here a 6.9V reference diode is used to bias Q1, a bipolar device, which allows operation up to the Q1 60V Vcb rating. The PN2907A is general purpose, but you can also substitute higher voltage parts, for example the 2SA1016. The 6.9V from D1 is reduced to about 6.3V across J1, which is sufficient to provide operation in the flat portion of the JFET device's output curve, where the rejection is highest. This is true at least for all devices of the J500 series, but some higher Vgs(off) individual JFETs could use even higher voltage bias (~10V).

Performance of this basic circuit is shown in **Fig. 9D** for two basic conditions: 1) a J507 operating at 1.5mA, and 2) a selected J113 JFET operating at 10mA. The J507 performance is generally very close to the noise level, except for a gradual departure at higher frequencies. The J113 is not quite as good, achieving

a LF rejection of 125dB, which would be equivalent to 1.8MΩ. This degradation in performance is typical of many current sources when operated at higher currents. Fortunately, there is a solution for it, which is the application of a cascode device for Q1, as originally discussed with **Fig. 6A**.

But, in terms of basic operation, some notes on D1 selection are appropriate here. This diode need not be a high precision part, and even ordinary 1N5230 series (and other similar) zeners will work if chosen for a ~6V breakdown at the final operating current. Note that this can lead to some pre-qualification for proper operating voltage for any ordinary zener; that is, one specified for such higher test currents as 20mA.

It is for this reason that an IC reference diode is preferred here, to alleviate this voltage uncertainty and allow predictable operation at 1-2mA. If an IC diode is used for D1, it can be the loosest tolerance of the family without impacting performance. The LM336Z-5.0 can also be useful in this circuit.

Note that this circuit has similarities

to **Fig. 6A**, but **Fig. 6A** depends upon the diode characteristics much more than this circuit. In this **Fig. 9C** circuit, the JFET (or other current source part) in the emitter of Q1 determines the output current, the stability, and so on.

Bruce Hofer, Chaiman and CoFounder of Audio Precision Inc., offered these comments on the tests:

"I would like to note that System One is now an obsolete product having been replaced in the mid 90s with our System Two, which itself has gone through several cycles of evolution. The technique of displaying noise spectrally using a swept 1/3-octave filter is still valid, but some engineers today would probably prefer to look at noise using the FFT. I should also note that analyzer analog input noise has improved (dropped) somewhat since System One, but only slightly. Indeed, many of our competitors today have yet to equal the performance of our original System One! Thus I think your graphs are still quite relevant."



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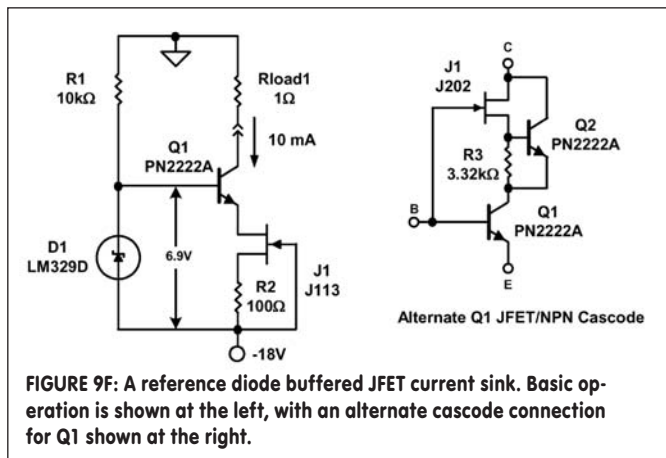
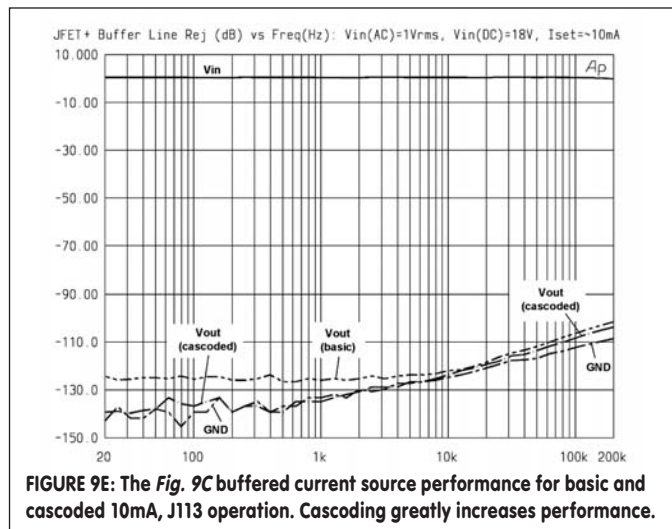
For high currents, a cascode connection can be used for Q1, and will reap performance benefits similar to those noted for Fig. 6A, operated with the simple cascode. But the simple cascode using a JFET such as the 2N5462 will be limited to currents of just 4mA within this circuit. To allow a greater degree of freedom in the cascode selection, you can use the alternate JFET/PNP cascode shown at the right in Fig. 9C for Q1. In this configuration, a

lower current JFET part (the 2N5461) is used, and is operated at 200µA. In typical operation, this biases the source of J1 about 1V below the gate, sufficient to drive Q2 at currents of 10mA or more, without saturation of Q1. In essence a composite JFET is formed, with a much greater freedom of operating current, but, importantly, still retaining the self-biasing feature of Q1. The alternate cascode at the right replaces Q1 in the circuit at the left, with the C, B,

and E terminals as noted.

The performance of this alternate cascode operating with the J113 at 10mA in the circuit of Fig. 9C is shown in Fig. 9E. Now the LF rejection is comparable to the noise level, and the HF performance is also improved. The use of this cascode is recommended for any current above a few mA, or whenever the high-end performance is required.

Reference Diode Buffered Current Sink
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sinks, and have similar options to those of **Fig. 9C** for cascoding, and so on. Because it is potentially confusing to say simply "use mirror image connections and complementary devices," a full current sink schematic example is shown in **Fig. 9F**. The basic circuit at the left is an exact complement to **Fig. 9C** operated at 10mA, with the J113 setting the current. In the case of this current sink circuit, biasing is from a -18V supply, and the output drives load Rload1.

Cascoding Q1 of this current sink can be done if a JFET/NPN setup is used, as shown in the option at the right of **Fig. 9F**. This alternate JFET/NPN cascode can be used at currents of 5mA and up, without concerns of JFET device pre-selection. This optional circuit operates similar to the alternate JFET/PNP cascode of **Fig. 9C**, but here uses complementary devices. To employ this cascode, connect the noted C, B, and E terminals within the left basic circuit at the three Q1 nodes.

NEXT TIME. . .

Part 2 will continue these discussions, and

will focus on higher current and higher voltage regulators with predictable, low TC output currents, and with very high rejection. It will conclude with some suggestions for applications and user modifications toward optimum use. **aX**

REFERENCES

1. Walt Jung, "Regulators for High Performance Audio, parts 1 and 2," *The Audio Amateur*, issues 1 and 2, 1995.
2. "LM134/234/334 3-Terminal Adjustable Current Sources," *National Semiconductor*, March 2005, www.national.com.
3. Arthur D. Evans, *Designing With Field-Effect Transistors*, McGraw-Hill, ISBN 0-07-057449-9, 1981.
4. "The FET Constant-Current Source/Limiter," Application Note AN103, Vishay/Siliconix, March 10, 1997, www.vishay.com.
5. "J500 Series Current Regulator Diodes," Vishay/Siliconix, July 2, 2001, www.vishay.com.
6. Selected and matched JFETs and JFET current regulator devices, as well as other audio components, are available from Borbely Audio. See www.borbelyaudio.com/audiophile_components.asp.



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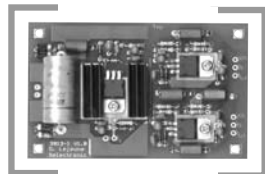
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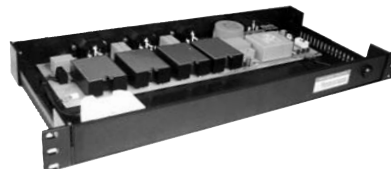
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Building a Center Channel for an Altec A7-800

By William Eckle

Here's a match made in speakerdom. This joint effort adds a center channel to a pair of classic Altec speakers.

About 20 years ago, a friend acquired a pair of Altec A7-800 (Voice of the Theater) speaker systems. He didn't like the cabinets and asked me to recommend an alternative. We decided an Altec "Malibu" cabinet would be the best compromise for size and performance. So we built a pair of thicker and better-braced copies of Altec "Malibu" cabinets and finished them in oak. My friend fell in love with these speakers over the years, both for their sound and appearance.

THE SETUP

Fast-forward to 2006. My friend now wanted a center channel speaker to go with his "Malibus." He specified he'd like to use Altec parts, if possible, and he'd like the center-channel speaker to match the general appearance (only much smaller) of his "Malibus," and be close in sound (timbre and so on) to his



PHOTO 2: The "Malibu" cabinet we built for the Altec A7-800.



PHOTO 1: Center channel (top) for existing Altec speakers. (Subwoofer is on right, out of picture.)

main speakers (Photo 1).

I wondered what Altec would offer in the way of a center channel speaker if they were still in business, to match the "Malibus" we had built (Photo 2). Having used Altec 8" 755s as TV speakers for years, I thought they'd make a good basis for a center-channel speaker, because they were designed for speech intelligibility, which is just what is needed for a dialog channel.

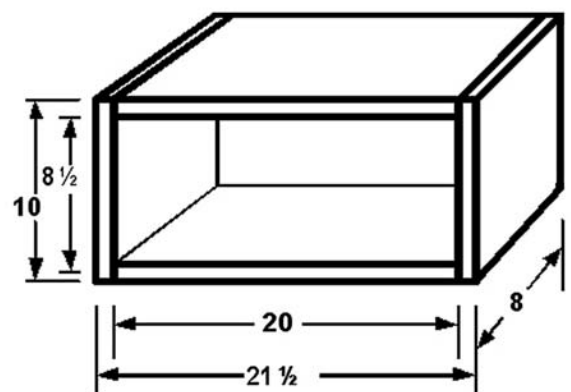
THE QUEST

We decided to look for a pair of Altec 8" 755 drivers and an Altec 3000H tweeter on eBay. Pairs of 755s were going for outrageous prices on eBay, but after 10 months and many failed bids, we acquired a pair of Altec 755s, a 3000H tweeter, and an N3000 crossover at reasonable prices in four separate purchases. It turned out that singles went for much lower prices than one-half-of-a-pair. We also purchased an L-pad from Madisound to adjust the tweeter level because the Altec crossover had a 0, -2, and -4dB tap to adjust the tweeter output. This would have been

awkward to adjust once the cabinet was sealed.

THE CABINET

We planned for the basic cabinet to be made from $\frac{3}{4}$ " Baltic birch (Fig. 1). Note: The front and back are $\frac{1}{4}$ " over-size to allow routing flush to the basic box once they are glued and screwed on the box. The addition of $\frac{1}{2}$ " MDF on the front is not only to deaden and strengthen the front, but also to allow later installation of $\frac{3}{4}$ " solid oak quarter-



3/4" Baltic Birch

Front & Back = $21\frac{3}{4}$ " x $10\frac{1}{4}$ " ea

Top & Bottom = 20" x 8" ea

Sides = 10" x 8" ea

1/2" MDF

Front = $20\frac{1}{2}$ " x 9"

Internal Volume: 1360 cu. in. (or 22.3 liters)

FIGURE 1:
Center channel
diagram and
specs.



PHOTO 3: Edge work on the center channel speaker.

round moldings on all the front edges.

We covered the entire box (including the bottom) with 1/4" oak plywood. I've found 1/4" plywood is easier to work with than veneer, and screws in the basic box don't need to be filled in. We routed the front to back and all back edges to accept 3/8" quarter-round, solid oak moldings (**Photo 3**).

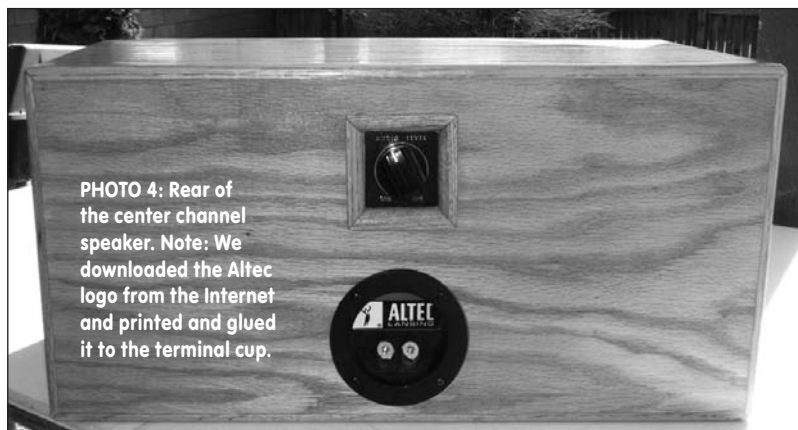


PHOTO 4: Rear of the center channel speaker. Note: We downloaded the Altec logo from the Internet and printed and glued it to the terminal cup.



PHOTO 5: Front of the center-channel speaker.

RESULTS

We applied sanding sealer and two coats of semi-gloss polyurethane, sanding between each coat. We used 20 ounces of "Fiber Fil" as stuffing.

The 755s are connected in parallel and are not connected to the crossover, so they run full range. The tweeter is connected to the 0Ω tap on the crossover for protection, and then to the L-pad for level adjustment.

We used several test discs and pink noise to check the balance of the system. The pink noise revealed a very close match across the front three speakers, after the levels were matched. Once my friend became used to having a center (dialog) speaker, he wouldn't be without it. He now runs all the TV audio through the main system, because he complains that the TV speakers are almost unintelligible.

I have a feeling the ghosts of Altec Lansing would be proud of this center channel speaker, as we are proud of accomplishing our goals of using vintage parts and close matching. *ax*

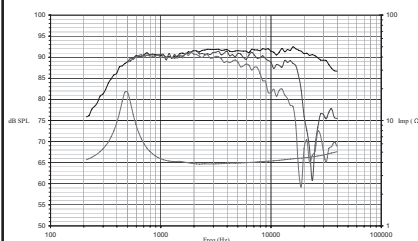
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- Cutout diameter: 77mm (3")
- Depth: 23mm (0.9")
- Flange thickness: 5mm (0.2")

Price Each: \$220.00 each



A Low-Noise Measurement Preamp

Gain of 1000 or 10,000, calibrated noise bandwidths including “A” weighting, and a low noise figure with low and high source resistances facilitate accurate measurements of noise and sub-microvolt signals.

By Dennis Colin

The preamp as shown in **Photos 1-5** is in a Hammond 1590C box (4.72" × 3.70" × 2.07"), which is tightly packed inside (not shown) with the circuitry on a solid copper ground plane board. I recommend the larger Hammond 1590D (7.38" × 4.70" × 2.05"), Digi-key HM154.

The circuit (**Fig. 1A**) uses a low-noise dual (paralleled) JFET (Q1), the Linear Systems LSK389B (see ads in *ax*) in a feedback configuration with the low-noise AD797AN op amp (U1) for the input stage. This provides very low input voltage noise (about 1nV/√Hz density,

148nV 20Hz – 20kHz noise, 112nV “A” weighted), negligible current noise, and very stable gain (100 = 40dB).

EQUIVALENT NOISE BANDWIDTH

Table 1 shows the noise bandwidth (NBW) of several common low and highpass filters. NBW means the LF or HF cutoff of an ideal “brick wall” filter that would, with white noise input (constant power per Hz BW), output the same total noise RMS voltage as the actual filter being considered.

The preamp has the following NBW selections:

1. LF points of 20Hz or 1kHz.



PHOTO 1: Low-noise preamp.

2. HF points of 20kHz or 100kHz.
3. “A” weighting: when used with the above NBW switches in the 20Hz –

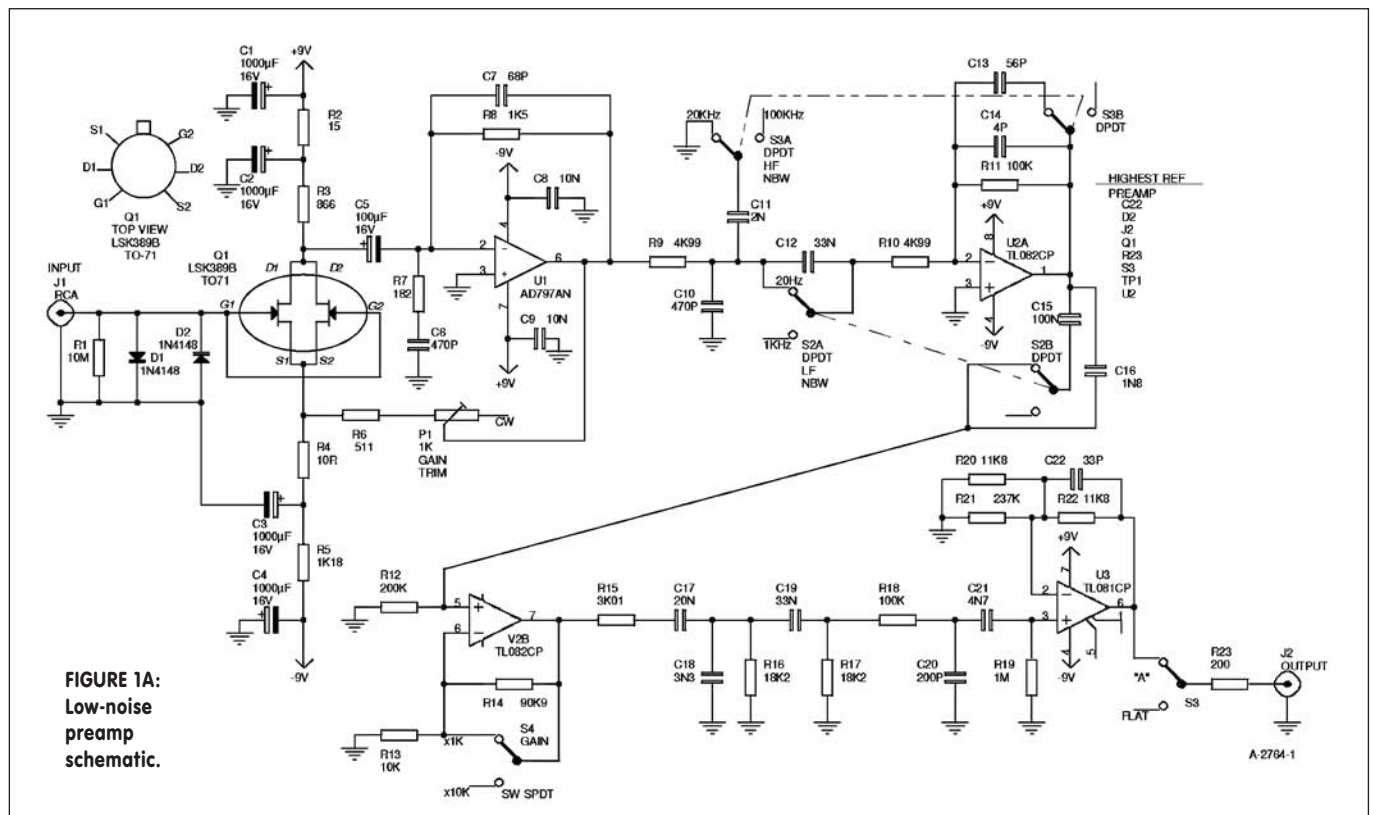
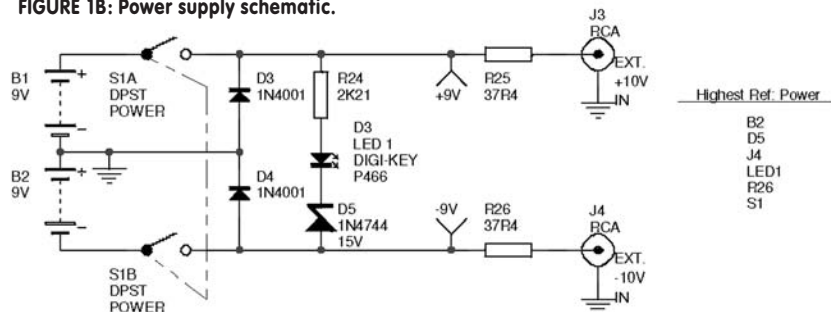


FIGURE 1B: Power supply schematic.



100kHz positions, the response is within ± 0.5 , -0.8 dB of the standard curve shown in Fig. 6.

The gain at 1kHz is within ± 0.1 dB of that with the "A" weighting switch off.

"A" weighting is used in signal/noise ratio (SNR) measurements because it's a good approximation to our low-level hearing response. The 1kHz LFNBW selection, like "A" weighting, greatly attenuates 60/120Hz AC line hum, but unlike the "A" curve, the pass band response is flat. Since 1kHz is only 5% of the audio band, the measurement (with fairly flat noise spectra) is close to 20Hz – 20kHz total noise voltage, and is useful

when there's significant hum contamination. The 100kHz HFNBW selection is provided when a flat audio response is needed, such as for spectrum analysis.

GAIN

With gain set to 1000 (60dB), microvolts in become millivolts out. With the preamps maximum output of ± 7.2 V peak, this allows a maximum sine wave of 5.1V RMS. But with typical (Gaussian) noise, 2.4V RMS is the maximum without significant error-producing peak clipping. So with gain of 1000, the maximum noise input is 2.4mV RMS.

The gain setting of 10,000 (80dB)

allows for the most sensitive noise measurements. The preamp's own "A" weighted input noise of 112nV RMS then outputs 1.12mV, which can be accurately measured on a true-RMS meter such as the Fluke 189 (\$500) or the much less expensive older Fluke 8010A. Noise levels lower than the preamp's own noise can be calculated by RMS subtracting it out.

GAIN STABILITY AND NBW VERIFICATION

I connected a precise 1000:1 attenuator to the input, consisting of a series 99.90k and shunt 100.00 Ω resistive divider, then set the gain with this attenuator to unity. The preamp's switches were set to gain = 1000, LFNBW = 20Hz, HFNBW = 100kHz, "A" weighting off. The preamp's gain at 1kHz was then set to 1000 ± 0.01 (± 0.0009 dB).

Gain change versus supply voltage was +0.037dB per volt. Temperature drift was -0.035 dB/ $^{\circ}$ C. Warmup change from 10 seconds to 10 minutes was -0.007 dB. Note: After power-on, it takes six seconds for the bias caps to sufficiently

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The frequency response plots in **Fig. 2** show the effects of the LF and HF noise bandwidth switches.

CIRCUIT

The input is DC-coupled to the JFET gates for lowest noise. D1, D2 protect the JFETs from static discharge, and so on. If an AC-coupled input is desired, I recommend a 100nF 630V film capacitor. I use one installed in a shielded cable assembly, for noise measurements on circuits (such as power supplies) having more than $\pm 100\text{mV}$ DC (the maximum allowable without a blocking cap). The series 100nF cap increases the preamp's 20Hz – 1kHz noise floor by about 0.4dB.

Input impedance is $3.6\text{M}\Omega/33\text{pF}$. This capacitance is low for a JFET input because the drain signal sees the “virtual ground” of U1's inverting input, thus eliminating Miller effect. R7, C6, and C7 ensure HF stability of U1 and the JFET/op amp feedback loop (U1 has a GBW product of 110MHz). R8 provides DC stability of U1.

For lowest noise and spurious pickup, R1, D1, D2, and C3 should have their ground terminals connected to a chassis ground lug on the input jack J1. The output of U1 has a gain of 100 (40dB) re the input, and a –3dB BW of about 10Hz to 500kHz.

SECOND STAGE

U2A has a gain of 10 (20dB). Switches S2 and S3 select the LF and HF rolloffs corresponding to the selected LF NBW

TABLE 1: Noise bandwidth of several non-resonant filters (multiple poles at the same frequency). NBW = equivalent noise cutoff frequency, low or high cut.

Filter Type	NBW/ pole frequency	dB at pole frequency	dB at equivalent noise cutoff	–3dB frequency/ noise cutoff
1st order LPF	$\pi/2 = 1.5708$	–3	–5.400	$2/\pi = 0.6366$
2nd order LPF (non-resonant)	$\pi/4 = 0.7854$	–6	–4.173	0.8178
3rd order LPF (non-resonant)	$3/16\pi = 0.5890$	–9	–3.881	0.8638
1st order HPF	$\pi/2 = 1.5708$	–3	–1.478	$2/\pi = 0.6366$
2nd order HPF (non-resonant)	$3/4\pi = 2.3562$	–6	–0.719	0.6608

TABLE 2: PREAMP INPUT NOISE.

NBW	INPUT REFERRED NOISE, RMS		
	Int. Batteries	Ext. AC/DC supply	Δ , AC/batt
“A” weighted	111.5nV	111.5nV	0.00dB
20Hz-20kHz	147.7nV, 1.045nV/ $\sqrt{\text{Hz}}$	149.4nV	0.10dB
1kHz-20kHz	134.0nV, 0.972nV/ $\sqrt{\text{Hz}}$	134.0nV	0.00dB
1kHz-100kHz	276.5nV, 0.879nV/ $\sqrt{\text{Hz}}$	276.5nV	0.00dB
20kHz-100kHz	241.9nV, 0.855nV/ $\sqrt{\text{Hz}}$	241.9nV	0.00dB
20Hz-1kHz	62.1nV, 1.984nV/ $\sqrt{\text{Hz}}$	66.1nV	0.54dB

and HF NBW. C15 and C16 also remove accumulated DC offsets.

THIRD STAGE

U2B has a flat response and selectable gain (by S4) of 1 or 10, for an overall preamp gain of 1000 or 10,000.

“A” WEIGHTING

The circuitry from U2B output to U3 output is based on that used by Charles Hansen in “Noise Meter Amp” (*aX* Jan. '05), but with higher impedances allowing smaller caps, and some tweaking. I

selected R21 for a 1kHz gain match ($\pm 0.02\text{dB}$) re the “flat” position of S5. Depending on “A” network component tolerances, you might need a different value for R21 (and possibly R20) if you want fractional dB accuracy.

POWER SUPPLY

Figure 1B shows the simple dual 9V alkaline battery supply used. The relatively high current drain of $\pm 21\text{mA}$ results in a battery life of about 10 hours. Of course, higher capacity NiCd, NiMH, or lithium ion/polymer batteries can be

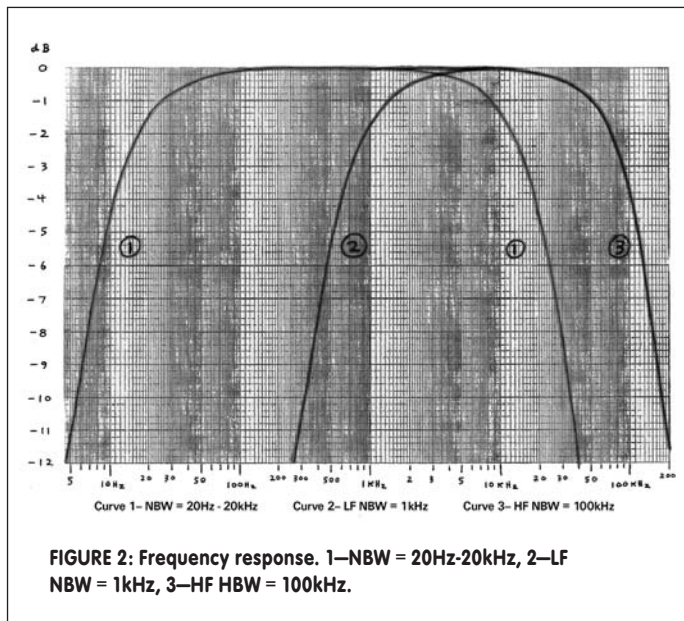


TABLE 3: PERFORMANCE DATA.

Gain = 1000 or 10,000 (60 or 80dB)
BW (–3dB) = 12.6Hz or 715Hz to 15.8kHz or 89.8kHz
Noise BW = 20Hz or 1kHz to 20kHz or 100kHz
Input noise voltage density = 1.045nV/ $\sqrt{\text{Hz}}$, 20Hz–20kHz NBW average 0.972nV/ $\sqrt{\text{Hz}}$, 1kHz–20kHz NBW average
Input Noise current density $\leq 0.1\text{pA}/\sqrt{\text{Hz}}$
Total input noise voltage, 20Hz–20kHz = 147.7nV RMS (–136.6dBV) same but “A” weighted = 111.5nV RMS (–139.0dBV)
Noise figure: 3.7dB (50 Ω), 1.2dB (200 Ω), 0.46dB (600 Ω), 0.28dB (1k), 0.05dB (10k)
Input impedance = $3.6\text{M}\Omega/33\text{pF}$
Output impedance = 200 Ω
Maximum input voltage: $\pm 20\text{mV}$ DC $\pm 100\text{mV}$ DC Gain = 1000: 5mV RMS sine, 2.4mV RMS Gaussian noise Gain = 10,000: 0.5mV RMS sine, 0.24mV RMS Gaussian noise
Maximum output voltage: 5.1V RMS sine, $\pm 7.2\text{V}$ peak, 2.4V RMS Gaussian noise
THD (mostly 2nd harmonic): approximately 0.002%
Battery supply: $\pm 9\text{V}$ at $\pm 21.2\text{mA}$
External DC supply: $\pm 10\text{V}$ at $\pm 21.2\text{mA}$, ground-isolated

used. The specified blue LED, with R24 and D5, serves as a low-battery indicator, extinguishing when the batteries discharge to 8.5V.

EXTERNAL SUPPLY

An AC line powered $\pm 10V$ supply can be used if it's ground-isolated. **Table 2** shows the preamp's input-referred noise with its batteries or an external LM317/LM337 regulated supply having about $40\mu V$ RMS integrated 20Hz – 20kHz noise. Note that the maximum preamp noise increase (0.54dB) is in the 20Hz – 1kHz band. With "A" weighting or a 1kHz LFNBW, the line-supplied preamp noise increase was less than 0.01dB.

An external $\pm 10V$ supply has another useful feature: while normally used with the battery switch S1 off, if this is on with the external DC supply, the battery voltages will charge, if needed, tending toward the $\pm 9.2V$ that is produced without batteries. This is completely safe, despite the "Do Not Charge" warning on 9V alkaline batteries. I've recharged these often over many years, provided that the current and voltage are safely limited, as they are here with regulated $\pm 10V$ applied. The equilibrium level of 9.2V is safely less than the 9.6V typical of a new "9V" alkaline battery. I haven't had one battery explode in over 15 years. But I know, "there's always a first time"—if you're concerned, keep S1 off when using external power.

POWER SUPPLY REJECTION RATIO

I injected 100mV RMS at 60 and 120Hz into the 10V supplies (using a transformer). With the preamp gain at 1000, and 20Hz – 20kHz NBW, the output ripple from the positive supply was 50.4mV (60Hz) and 13.0mV (120Hz). From the negative supply, the output ripple was 7.3mV (60Hz) and 1.0mV (120Hz). The op amps have excellent PSRR, but not the single-ended JFET input!

The worst-case PSRR (positive supply at 60Hz) is thus about 2, or 6dB, re the output with gain of 1000. This agrees with the lowest row in **Table 2**, where RMS subtracting the 20Hz – 1kHz noise voltages yields 22.6nV RMS input-referred line ripple. Not bad for an AC line powered supply; 22.6nV is 194.5dB below the all-surrounding 120V AC. 194dB is the brightness ratio of two

lightbulbs, one at 3" from your eyes, the other one at the distance of the moon!

WHAT'S WRONG WITH CONVENTIONAL NOISE MEASUREMENTS?

Nothing's wrong with the numbers plotting noise density (e.g., nV/ $\sqrt{Hz}$) versus frequency. The problem is that such plots don't express the *perceived* level and spectrum, because of three factors:

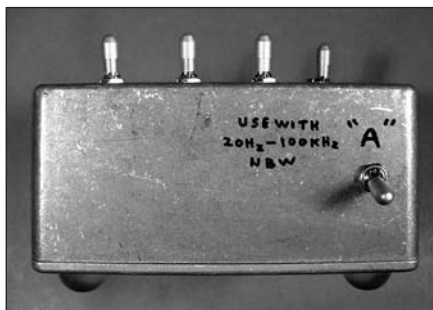


PHOTO 2: Front.

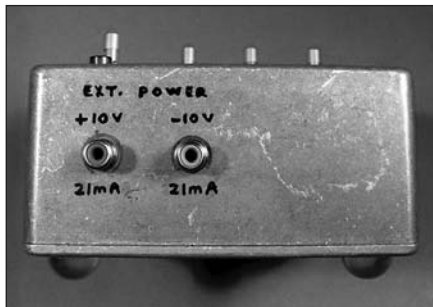


PHOTO 3: Rear.

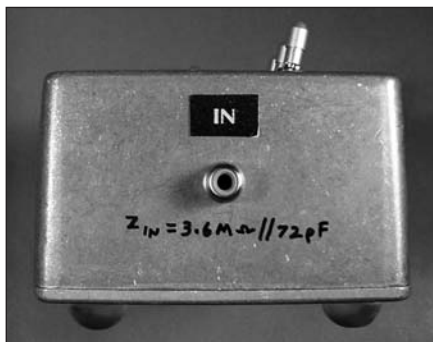


PHOTO 4: Left side.



PHOTO 5: Right side.

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1. Conventional spectrum analysis uses a constant bandwidth; $nV/\sqrt{\text{Hz}}$ means the RMS noise in a 1Hz BW. On such a graph, white noise is flat. However, you don't hear this way, but rather on a log frequency basis; e.g., per octave, $\frac{1}{3}$ octave, and so on. On such a spectrum graph, white noise has a constant upward 3dB/octave slope, and *sounds* "hissy".

That's why *pink noise*—with its con-

stant power per octave sounding like a steady ocean wave noise—is used. On a constant BW plot, pink noise (sonically misleadingly) has a constant downward 3dB/octave slope.

So, the reasonably "white" spectrum of many audio components (similar to this article's preamp), fairly flat on a per-Hz basis above 1kHz, has that "hissy" sound, heard mostly in the higher frequencies where the ear is

most sensitive at the (you hope) quiet levels of component noise.

2. A linear dB (log voltage) vertical scale would be in accordance with your hearing.
3. The "A" weighting curve (for low-level sounds) further emphasizes (perceptually) the mid-treble range; the "A" curve peaks at 2.5kHz, like your low-level hearing (**Fig. 6**).

For phono preamp design, the audibility of the amplified cartridge plus preamp input noise is modified by a fourth factor, the RIAA curve. Fortunately, this strongly attenuates the higher frequencies.

NOISE SPECTRA OF THIS PREAMP, WITH AND WITHOUT THE ABOVE FACTORS

Figure 3 shows the unweighted input noise density spectrum. Curve 1 is conventionally plotted, using $nV/\sqrt{\text{Hz}}$. Curve 2 shows the same data, but with constant per-octave analysis, the way you hear. Above 1kHz the $nV/\sqrt{\text{Hz}}$ density is flat within 3.1dB p-p (reasonably "white"); the per-octave curve reflects this as an exponential rise.

The unweighted noise curve in **Fig. 4** shows this as an upward slope approaching 3dB/octave from 1-20kHz. But applying "A" weighting (for low-level sounds) with the dB scale of **Fig. 4** shows the noise spectrum as you hear it. It broadly peaks around 6kHz. Note that "A" weighting reduces the integrated 20Hz – 20kHz noise level by only 2.4dB, whereas some components might show an 8dB or so reduction.

This has nothing to do with the per-octave analysis; noise power is noise power no matter how it's (accurately) measured. To see why the reduction here is only 2.4dB involves the following:

1. The "raw" ($nV/\sqrt{\text{Hz}}$) spectrum in **Fig. 3** curve 1 has a strong LF rise below 1kHz. With the log frequency scale, this appears to affect half the spectrum. But 1kHz is only 5% of the 20kHz audio (and noise) BW. That's why the audio band averaged noise density is only $1.045nV/\sqrt{\text{Hz}}$, about the level at 2kHz. The "A" curve mostly attenuates below 1kHz (**Fig. 6**), again affecting only 5% of the audio band.
2. Notice in **Fig. 6** that at the peak frequency of 2.5kHz, the "A" curve is

TABLE 4: PARTS LIST.

Reference	Description	Size	Mfr.	P/N	Distributor	P/N
C14	cap, 14pF					
C22	cap, 33pF					
C13	cap, 56pF					
C7	cap, 68pF					
C20	cap, 200pF					
C6, 10	cap, 470pF					
C16	cap, 1.8nF, film	100V			DK	PS1182J
C11	cap, 2.0nF, film	100V			DK	PS1202J
C18	cap, 3.3nF, film	100V			DK	PS1332J
C21	cap, 4.7nF, film	100V			DK	PS1472J
C8, 9	cap, 10nF, film	100V			DK	PS1103J
C17	cap, 20nF	100V			DK	PS1203J
C12, 19	cap, 33nF, film	100V			DK	PS1333J
C15	cap, 100nF, film	100V			DK	PS1104J
C5	cap, 100μF	16V				
C1, 2, 4	cap, 1000μF	16V				
C3	cap, 2200μF	16V				
D1, 2	diode			1N4148		
D3, 4	diode			1N4001		
D5	Zener, 15V, 1W			1N4744A	DK	1N4744ADICT
J1, 4	phono jack, green				DK	CP-1417
J2, 3	phono jack, red				DK	CP-1413
LED1	LED, blue				DK	P466
R4	res, 1%MF, 10Ro	$\frac{1}{4}W$				
R2	res, 15Ro	$\frac{1}{4}W$				
R25, 26	res, 37R4	$\frac{1}{4}W$				
R7	res, 182R	$\frac{1}{4}W$				
R23	res, 200R	$\frac{1}{4}W$				
R6	res, 511R	$\frac{1}{4}W$				
R3	res, 866R	$\frac{1}{4}W$				
R5	res, 1K18	$\frac{1}{4}W$				
R24	res, 2k21	$\frac{1}{4}W$				
R15	res, 3k01	$\frac{1}{4}W$				
R9, 10	res, 4k99	$\frac{1}{4}W$				
R13	res, 10ko	$\frac{1}{4}W$				
R20, 22	res, 11k8	$\frac{1}{4}W$				
R16, 17	res, 18k2	$\frac{1}{4}W$				
R8	res, 51k1	$\frac{1}{4}W$				
R14	res, 90k9	$\frac{1}{4}W$				
R11, 18	res, 100k	$\frac{1}{4}W$				
R12	Res, 1% MF, 200k	$\frac{1}{4}W$				
R21	res, 237k	$\frac{1}{4}W$				
R19	res, 1M00	$\frac{1}{4}W$				
R1	Res, 10M, 5%	$\frac{1}{4}W$				
S1-5	switch, DPDT, lock				DK	450-1487
TP1	trimpot, 1K				DK	3296Y-102
Q1	dual JFET		LS	LSK389B T0-71		
U1	op amp		AD	AD797 AN	DK	AD797AN
U2	dual op amp			TL082CP	DK	296-1780-5
U3	op amp			TL081CP	DK	296-7203-5
	box		Hammond	1590D	DK	HM154
	LED lens, blue				DK	L3006
	battery, 9V alkaline (2)					
	9V battery holder (2)				DK	BH9V-W
	8 pin DIP socket (4)				DK	A9408
	rubber bumper (4)		SPC	2567	Newark	92N4782

DK = Digi-Key, LS = Linear Systems, AD = Analog Devices

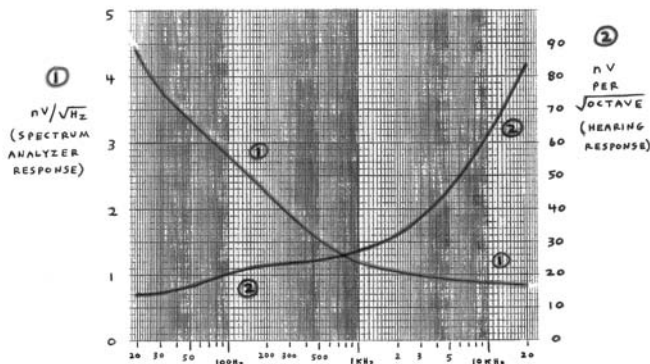


FIGURE 3: Input noise density spectrum. Integrated 20Hz-20kHz = 147.7nV RMS = -136.6dBV. 1-nV/√Hz (spectrum analyzer response), 2-nV per √octave (hearing response).

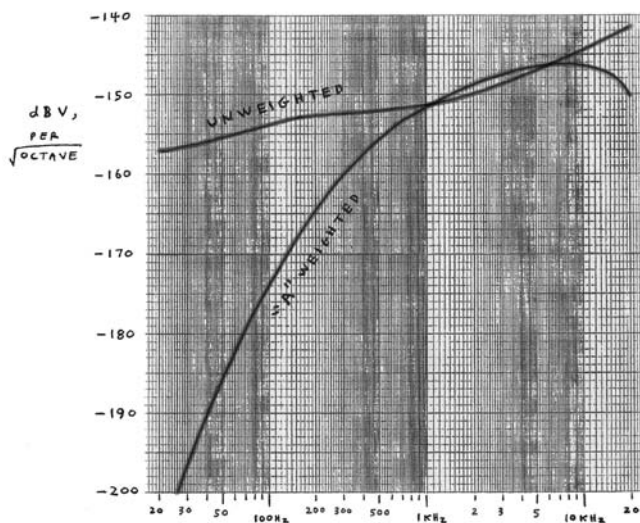


FIGURE 4: Input noise, per-octave basis. Unweighted (1) 20Hz-20kHz = 147.7nV RMS = -136.6dBV. "A" weighted (2) 20Hz-20kHz = 111.5nV RMS = -139.0dBV Δ = -2.4dB.

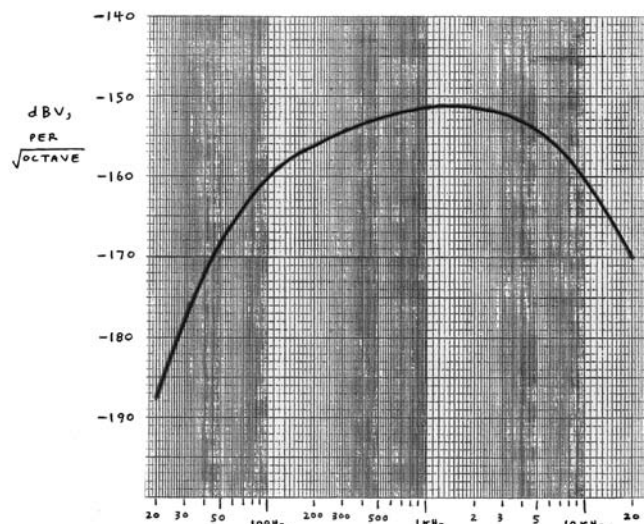


FIGURE 5: Input noise, per-octave basis, with "A" weighting and RIAA curve. 20Hz-20kHz noise = 67.4nV RMS = -143.4dBV Δ = -6.8dB re unweighted and no RIAA curve.

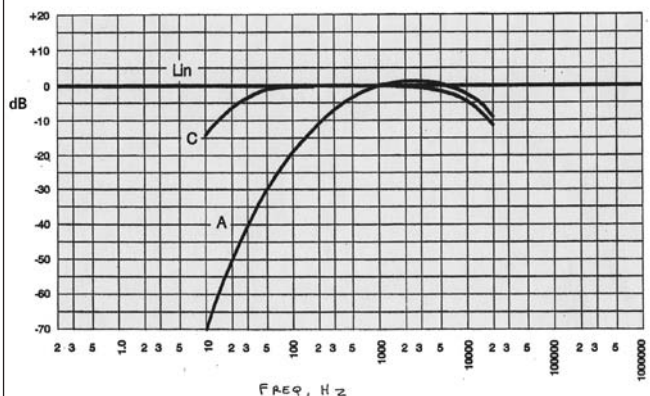


FIGURE 6: "A" and "C" weighting curves.

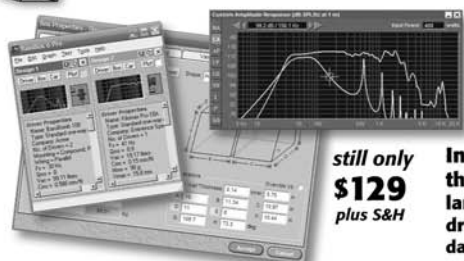


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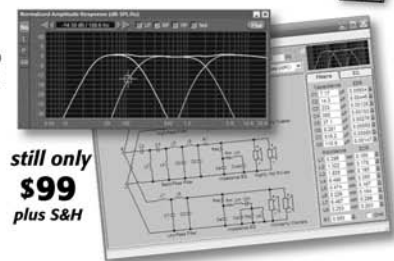
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by Mark Sanfilippo

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1.1dB higher than the flat ("Lin") reference for loud sounds. Apparently, this reflects the ear's maintaining high sensitivity from 1-5kHz down to the quietest audible SPL. In any case, with noise having strong 1-5kHz content (such as white or near-white noise), you can see that the "A" weighting curve doesn't reduce the overall noise level much.

- Often, the significant "A" weighting reduction of a component's "noise" is due to that "noise" containing significant 60/120Hz power line hum (and harmonics). When you're considering nanovolts of noise per Hz BW, it doesn't take much discrete-frequency hum to dominate the raw measured noise + hum power. But, of course, the hum, being of low frequency and (you hope) low level compared to the audio signal, is greatly attenuated by the "A" curve (28dB at 60Hz, 17dB at 120Hz) and by your low-level hearing.

THE RIAA CURVE INFLUENCE

Figure 5 shows the preamp's input noise spectrum when all four appropriate factors are considered: per-octave hearing, linear dB (logarithmic) hearing, the "A" curve, and the downward-sloping RIAA curve. Note that here the noise has a broad peak around 1.5kHz, with strong LF and HF attenuation. Consequently, the perceived noise reduction is 6.8dB with regard to the raw input noise.

Therefore, with a typical near-white input noise distribution above 1kHz, the phono preamp's output SNR will be about 6.8dB better than the raw input noise level would indicate. Phono preamp designers, remember that number! (I just built one, the "LP797"; that num-

ber is close.)

For non-RIAA components, the SNR improvement is about 2.4dB.

MEASURING PERCEIVED NOISE LEVEL

- Connect the preamp to the component or circuit under test with, of course, shielded cable. Because the preamp input is DC-coupled, make sure the tested device has no more than $\pm 300\text{mV}$ DC offset. (You can use a series 100NF blocking cap, but this increases preamp LF noise by about 0.4dB.)
- Connect the preamp's output to a true-RMS AC voltmeter capable of reading down to at least 1mV. (With less sensitive meters you can use a post-amp, provided it's flat from 20Hz to 40kHz.)
- Set the preamp's switches to 20Hz LFNBW, 20kHz HFNBW, and gain to 1000.
- If the meter reads less than 200mV RMS, you can increase the gain to 10,000 without saturating the preamp's output.
- Divide the measured voltage by the preamp gain; this is the component's unweighted 20Hz – 20kHz noise output.
- Switch the "A" weighting on, and set the HFNBW to 100kHz (this makes the response flat to 20kHz before the "A" filter, ensuring accurate "A" response). Divide the measured voltage by the preamp gain; this is the component's "A" weighted noise. The ratio of the component's maximum undistorted signal output to this noise in dB (20 times the log of this ratio) is the "A" weighted dynamic range. Using the component's nominal signal output

TABLE 5: MEASUREMENT POWER SUPPLY NOISE.

Power Supply	Vo io	RMS Output Noise		
		20Hz to 1kHz	1kHz to 20kHz	20Hz to 20kHz
LM317 $C_{IN} = 2\text{mF}$, $C_{ADJ} = 0$, $C_O = 10\mu\text{F}$	+10V 32mA	65 μV	206 μV	216 μV
LM337 $C_{IN} = 2\text{mF}$, $C_{ADJ} = 0$, $C_O = 10\mu\text{F}$	-10V 32mA	74 μV	202 μV	215 μV
LM317 $C_{IN} = 10\text{mF}$, $C_{ADJ} = 10\mu\text{F}$, $C_O = 10\text{mF}$	+10V 32mA	40.0 μV	19.6 μV	44.6 μV
LM337 $C_{IN} = 10\text{mF}$, $C_{ADJ} = 10\mu\text{F}$, $C_O = 10\text{mF}$	-10V 32mA	18.5 μV	25.2 μV	31.3 μV
Leader LPS - 152	+12V 200mA	38.4 μV	7.6 μV	39.2 μV
Marlin P. Jones 1590PS	+30V 7.3A	295 μV	78 μV	305 μV
Duracell 9V Alkaline Batteries				
MN1604 6LR61	+9V 9mA	<10nV	<10nV	<10nV

The AD797AN is available from Digi-Key (\$8.66) and Newark (\$8.84)

The LSK389B T0-71 is available from Linear Integrated Systems, (510) 490-9160

(e.g., 1V RMS, 0.775V RMS, and so on) results in the "A" weighted SNR.

VERY LOW NOISE MEASUREMENTS

With any component having at least 20dB gain, its amplified input noise will override this preamp's input noise, so that the preamp won't significantly contaminate the measurement. The preamp's 20Hz – 20kHz integrated noise is 147.7nV RMS on the proto unit, but it's subject to the LSK389B noise. The datasheet says 0.9nV/√Hz typical, 2.5 maximum. I recommend using a DIP socket for Q1 so you can screen several JFETs for lowest noise.

In any case, noise levels below the preamp's own noise can be calculated by RMS subtraction:

1. Measure the preamp output noise while connected to the component under test.
2. Connect a shielded and shorted RCA plug to the preamp input, and again measure its output noise.
3. RMS subtract the two readings. For example, suppose the output with the component is 1.935mV, and with the input grounded it's 1.477mV (the proto unit with gain of 10,000).

Then square the 1.935, subtract 1.477 squared, and take the square root of the result: 1.250mV. Dividing this corrected output by the gain of 10,000 (used in this example) gives the component's noise, 125nV RMS, corrected for the preamp's own noise.

POWER SUPPLY NOISE MEASUREMENTS

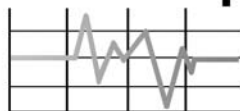
Table 5 shows the noise of several DC power supplies, using the 100nF blocking cap cable. Noise from 20kHz – 100kHz (not shown) was much lower than audio band noise, except for the Marlin P. Jones unit (a 30V, 20A switching supply); switching at 100kHz, the 20kHz – 100kHz noise is 670μV RMS.

The noise of the 9V alkaline battery was unmeasurable (with this preamp); it was no greater than the uncertainty of 10nV RMS. *aX*

REFERENCE

1. Charles Hansen, "Noise Meter Amp," *aX* Jan. '05.

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Speech Intelligibility, Part 2

What reverb is... and how to calculate it.

By Ed Simon

In last month's Part 1, we identified noise as a problem to good sound and examined ways in which we could control it. The second impediment to good listening or speech recognition is the self-masking of the sound by itself. Before the advent of modern acoustics, this was sometimes called resonance, or reverb. In today's politically correct atmosphere, you need to be careful about what you call things.

An echo is a single clean reflection of a sound. Reverb is a stream of multiple echoes, decaying into an indistinguishable muddle of noise. Resonance is a single frequency or band of frequencies that decay at a very slow rate. If the energy being fed into the resonance is greater than the decay, this results in a tone. A canyon echoes, a room reverberates, an organ pipe resonates.

ACOUSTICS LESSON

The prediction of decay in a room was the first major step in the modern practice of acoustics. This started at Harvard University where Wallace Clements Sabine, a junior faculty member, was asked to investigate why a new lecture hall was not performing as expected.

He started with a pair of lecture halls, an assortment of organ pipes (C"), a stopwatch, compressed air, and seat cushions. He sounded a pipe until the volume in the room became very close to steady-state, then he turned it off and timed how long the sound took to decay. He tried single organ pipes and multiple pipes of the same tuning for increased amplitude in a variety of rooms.

Sabine devised a formula for predicting the decay of sound based upon the volume of a room and the total absorption in it. He found that the decay of sound increased as he added absorption. It did not seem to vary with increases in starting level. It took a bit of time for the sound energy to rise to an almost steady level. Using a decay ratio of 1,000,000 to

1, he came up with this classic equation:

Time to decay $1,000,000/1 = .16 \times \text{Room Volume/Total Absorption}$.

For those in the know, that would be right! He used the metric system of measurements.

In some early experiments Sabine instructed a panel of experts to listen to a pianist while he added seat cushions to the rooms. All the rooms were furnished differently and varied in size. By adjusting the added absorption, Sabine was able to determine that the preferred reverb time occurred at just about one second. He was forced to measure the reverb time with organ pipes and a stopwatch at night when no one else was present and it was quiet enough.

These and other experiments make it clear that Sabine changed the science of observing the behavior of sound into the engineering practice of using knowledge to predict and design. He published his first work in the periodical *The American Architect* (1900).

VOLUME

There were a few changes that followed in this country. The English system of measurements changed the formula to $T = .05 \times V/A$. Absorption is now measured in sabins. One sabin of absorption requires that all the sound is absorbed in an area of 1ft^2 (or meter.) Zero sabins means no sound is absorbed, it is all reflected. A ratio of 1,000,000 to 1 is now referred to as 60dB.

Sabine understood that it takes time for the energy to build up in a room and then decay. He determined that decay was based on the absorption and geometry of the room and knew that as more people entered the room they absorbed some of the energy. He was the first to define the difference between reverb and resonance and established that



PHOTO 1:
Sample absorber.

under normal conditions reverb time is not level-dependent. He even determined that the optimum reverb time for a given room was volume-dependent. He suggested 1 second for $10,000\text{ft}^3$ and 2 seconds for $1,000,000\text{ft}^3$. In a large room the extra reflected sound helps to naturally amplify speech and music.

Sometimes you need to calculate the reverb time for a room before it is built or renovated. This is reasonably easy to do. Remember the reverb time for many rooms is $.05 \times \text{volume/absorption}$. You must be able to calculate the volume of the room and the total absorption. For a rectangular room the volume, of course, is the height $\times$ the width $\times$ the length.

ABSORPTION

To calculate the absorption, note what the walls, floor, and ceiling are made of. Then consult a chart to find out the absorption of that material. The chart will show the absorption for different frequency ranges. If you are just considering speech, use the 2000Hz data only. Sabine used C", about 512Hz; we still use 500Hz for music rooms. If you are careful, prepare a calculation for each frequency.

The values for absorption originally were determined by applying sound energy perpendicular to the material and measuring the absorption. But the bad news is that sound in a room does not always hit dead on. That is why the currently accepted method is to place much of the material in a reverb chamber and measure how it decreases the reverb time, although this gives a lower value

because not all of the energy hits dead on. The error here is that the average angle in a reverb chamber is still greater than the average angle in a large room. It is just about right for listening rooms or classrooms but frequently high for cathedrals.

If you are considering cathedrals or stadia, a more accurate method is to use the highest absorption value and then calculate the average angle at which the sound strikes the material. Then multiply the cosine of this angle squared by the highest absorption value to get the value for that angle.

Another phenomenon is that the same amount of absorption spread out over a room will reduce the reverb time more than if it is all clumped together. Some of this may be due to higher frequencies propagating as waves, whereas the lower frequencies inside a room do not have enough length to form a wave so they behave as pressure modulations. The higher frequency waves hitting the spread-out material may have some phase shift induced by the slower propagation near the absorbent material, resulting in additional energy loss. The lower frequency pressure modulations are perhaps guided by the untreated area into the absorption so the increase in edge area increases the total absorption.

AIR COUNTS

Table 1 is a chart of typical absorption values, which differs from the numbers you may find other places. I started with other folks' numbers and sent my people to measure larger rooms and compare the predictions with the measured results. I kept notes, and this table is the

result.

Take the floor, for example. If it is carpet, determine the thickness, pick the value, and multiply that by the area of the carpet. That is the absorption for the carpet in the room. If parts of the floor are concrete, use the value(s) for concrete times the area. Concrete block is a common building material with a giant problem!

When I attended college, the reasonably new recitation rooms were concrete block rooms with a concrete floor and ceiling. There were no windows, because electricity was considered reliable enough to provide all the light required. The concrete block walls and student bodies provided enough absorption so that speech intelligibility was good.

In those ancient days material was presented with chalk on a real blackboard! The problem was that the erasers for the blackboards would fill with chalk dust and thus needed to be cleaned. In a classroom without windows the solution was to pound the erasers against the wall leaving white marks. Eventually the walls needed to be painted, thus lowering the sound absorption in the rooms to almost zero! Speech intelligibility dropped to awful. Now the fix was to add acoustic ceilings, but to save money, they used cheap but more durable tiles. There was just enough absorption to bring the rooms up to adequate.

The moral of this tale is that when designing with concrete block as the absorption material, sooner or later it will be painted.

Be sure that you total all of the surface areas with their values. Add to this total of the boundary absorption your

best guess for the padded furniture and people in the room. This grand total is the number you use in the equation. In big rooms air counts!

A quicker, simpler method is to assume everything absorbs zero except for carpet, furniture, drapes, and other fuzzy stuff—these have a value of one. Your absorption value will be the surface area of all your fuzzy stuff.

Once you have the volume and absorption for the room, calculate:

$$.05 \times \text{Volume/Absorption}$$

The number should be more than zero and less than ten.

The biggest problem with the estimate is that the Sabine equation predicts a reverb time even for a room in which everything is absorbent. The equation works for most rooms, but probably not for your listening room.

BOUNCIN' ROUND THE ROOM

The method for calculating the reverb time of rooms that are not very live is the Norris-Eyring equation. For the Sabine equation you needed to know the volume of the room and the total absorption.

TABLE 1.	125	250	500	1000	2000	4000
Brick	0.02	0.02	0.03	0.03	0.04	0.06
painted brick or concrete	0.01	0.01	0.01	0.02	0.02	0.02
3/8" carpet on concrete	0.01	0.02	0.04	0.07	0.15	0.29
3/4" carpet or carpet + pad on concrete	0.02	0.04	0.07	0.15	0.29	0.58
3/4" carpet or carpet + pad on wood	0.16	0.12	0.13	0.18	0.32	0.61
concrete block new unpainted	0.5	0.45	0.42	0.41	0.29	0.25
concrete block painted 1-2 coats	0.22	0.12	0.08	0.09	0.09	0.08
plaster	0.03	0.02	0.02	0.04	0.06	0.07
stained glass windows	0.24	0.09	0.07	0.06	0.05	0.03
drywall	0.22	0.1	0.05	0.04	0.05	0.08
light drapes	0.05	0.15	0.25	0.45	0.5	0.5
thick drapes	0.15	0.25	0.45	0.5	0.55	0.6
audience summer	2.4	3.4	3.8	4.2	4.5	4.2
audience winter	4.2	4.8	5	5.8	6.1	6.3
pew cushion bottom only per linear foot	0.4	0.65	0.8	0.86	0.86	0.8
pew cushion bottom and back	0.8	1.2	1.5	1.6	1.7	1.5
wood seating per person empty	0.68	0.55	0.47	0.32	0.28	0.24

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the appearance of your room, try hiding the fuzz. The underneath of a sofa is often hollow. If you put a bag of fuzz here, only the high-frequency energy that bounces off the carpet would be absorbed. More of the midrange would get in and the very lows would pass through. A taut fabric hidden here would remove some of the low-frequency energy.

The more useful trick is to control the midrange without killing the high frequencies: Diaphragms for low frequencies, fuzz

for highs, and a good balance to get the midrange right. You should already know to take the grille cloth off your loudspeakers for best results. If you think it is the fuzz in the cloth that kills the highs, that is only part of the answer. Uniform holes act as filters. The holes' size and spacing determine the frequencies of interest.

You can construct a baffle of a thin surface material with holes in it backed by fuzz to absorb midrange energy. The high frequencies will penetrate. To a small degree most will be bounced back by the surface material. At just the right frequencies some will pass to the fuzz, some will bounce, and some will become twisted in phase on the bounce and cancel out a bit. A well-designed absorber will have some diaphragm action for lows, holes for midrange, and fuzz for highs.

ABSORBER CONSTRUCTION

As an example, in churches we use an absorber that is 20" x 60" x 6" made from 1/8" birch plywood, all painted black. There are 888 15/16" holes in the face of the box, and the back is open. It is stuffed with 8" of fiberglass compressed into 6", and the face and sides are wrapped in a grille cloth. With a white grille cloth, we use a black fabric to cover the face to keep the holes invisible. These absorbers hang a few inches down from the ceiling, allowing them two chances at the sound energy.

You can make a simple corner absorber

from fiberglass hardware cloth and fabric. Purchase a galvanized steel wire mesh called hardware cloth or screen from your hardware supplier. This is woven from thin steel wire that has been hot-dip-galvanized and forms a mesh with 1/4" open-

ings. This is sold in 3' widths and is available by the roll or cut to length. Six to 8' is usually enough to make one absorber.

I use 78" because that length fits nicely in my car. When I cut the hardware cloth, I wear leather gloves and use aircraft

Table 3: You can use this worksheet to determine absorption.

	Quantity	Height	Peak	Width	Length	Material	Area	Corrected Area	ABSORPTION					
									125	250	500	1000	2000	4000
Front Wall							0	0						
Windows in Front Wall							0	0						
Doors In Front Wall							0	0						
Other														
Rear Wall							0	0						
Windows in Rear Wall							0	0						
Doors In Rear Wall							0	0						
Other														
Left Wall							0	0						
Windows in Left Wall							0	0						
Doors in Left wall							0	0						
Other														
Right Wall							0	0						
Windows in Right Wall							0	0						
Doors in Right wall							0	0						
Other														
Ceiling							0	0						
Other														
Floor							0	0						
Isle (s)							0	0						
Isle (s)							0	0						
Pews														
Other														
Other														
People														
Minimum														
Maximum														
Other														
Other														

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straight snips and try to cut in the center of the cell. Then I use needlenose pliers to bend the sharp edges back over the end wire and press down the sharp tips.

Use an un-faced R25 8" thick × 15" wide fiberglass. Put on your latex gloves and safety glasses. Cut two pieces almost the length of your hardware cloth. I use 72". You can place this about 2" in from the long edge of the hardware cloth and roll it closed.

Try to use cable ties to hold it together until you solder the seam. Be sure the hardware cloth cells line up with each other or else your absorber will tilt when you are done. I count four cells in and two down to start with and cable-tie both ends. Then I add about six more ties carefully spaced and pull the seam almost together.

Solder the seam using a 40W iron and 60/40 rosin core .032" solder. The seam wires should just touch as you solder them. Apply just a drop of solder every 4" for about a 24" length. Then solder a bead from drop to drop. After it cools, fill the small gap and go on to the next section.

I just use black duct tape to attach the black fabric covering. Use a 36" wide cloth that is intended for gardening, but if you are using these in a public space you should use a fire-resistant cloth. You could also wrap the absorber with a nice-looking piece of fabric that matches your décor and sew the seam. Fold over the top and bottom and join them in the center.

You should place the absorber in a corner about 4" from each wall. The walls and spacing make it more effective at low frequencies.

Some people use a Helmholtz resonator to absorb specific frequencies. In one practical application it resembles a large flowerpot partially filled. Think of a small indoor tree in an over-large pot with a narrow tapered top one-third filled with dirt. Adjust the topsoil to tune it.

PLAYING THE ROOM

The best classic reverb time for speech intelligibility in a small room is about one-half of a second. Any less and too much of the speech is absorbed also. Any longer and the intelligibility scores begin to drop. The horrible news is that if you make the room optimum for speech, then it is too dead for music. The good news is that you do not need the maximum speech intel-

ligibility in most cases. You just need good enough.

In a concert hall you can live with the high end of acceptable reverb times, because the music is more important. In a multipurpose hall, you can vary the absorption. That way when a play comes in, the room will be dead.

Speech intelligibility is at maximum once the signal-to-noise-ratio in a room reaches 20dB. This signal-to-noise might work for heavy metal, but the symphony crowd would write you very nasty notes.

A room that is really bad for speech also hinders music. Even though many organists yearn for very long reverb times to increase the power of the music, if it is not properly controlled, all the notes just smear together in a muddle. The definition of the music is lost.

The reverberant field theory that most folks work with assumes a clean uniform decay without sharp echoes or resonances that distort the decay of the energy in the room. A room's reverb is called linear if the decay when plotted on graph paper is a straight line. The graph paper should have the vertical (Y) axis as a six decade log scale to represent the 60dB of decay. The horizontal (X) axis should be linear. When the plotted curve wiggles a bit, the room is nonlinear. This is often caused by the room's geometry, materials, and construction.

I prefer multipurpose rooms that are nonlinear. A sharp drop of 10dB followed by a gradual rolloff can give a classic reverb time of several seconds for the music and still have excellent speech intelligibility.

A good room will allow you to hear the first sounding of a note and still support the music. That is why I prefer nonlinear rooms. The quick drop of 5-10dB in the reverb time curve will give detail to the music and allow speech intelligibility. The tail can extend for several seconds giving a longer classic reverb time. The exact drop depends on the primary use of the room.

For more modern music a reverb time of 2.5 to 3 seconds may be more appropriate. This is where Art sings volumes. You can judge when your adjustments to your room are right by your listening tests or measurements.

In next month's installment, we'll look at the problems associated with reverb. *ax*

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A Flexible Subwoofer Amplifier, Part 4

Adjustments, Testing, and New Ideas.

By Rudy Godmaire

Parts 1-3 explained the design and the making of my subwoofer amp. In this last part, I will discuss its performance and share some measurements, tuning procedures, and listening impressions. As a “bonus feature” (see sidebar), I will give some hints on how to make two very nice preamps using the same PCBs.

OVERVIEW OF THE PERFORMANCE

Figure 1 summarizes the behavior of the subwoofer amp with a dummy load of 5Ω . Using a test disk (*Stereophile* Test CD 3, track 17) and an input signal of 0.5V RMS in each stereo input of the subwoofer amp, I generated various frequencies and measured the output voltages at the binding posts. The results are shown for two cutoff frequencies f_{CLP} : 48Hz and 80Hz.

It is interesting to compare the simplified theoretical slopes of attenuation represented by the thin dotted lines, and the real ones depicted with the bold lines. Though the real attenuation slopes begin to drop sooner than I expected,

they nevertheless cross the -18dB point at almost the same place as the theoretical slopes. This graph clearly shows that the third-order low-pass Butterworth active filter does its job as intended.

These results with a dummy load are very satisfying. However, what interests me is the behavior in real-world situations. **Figure 2** presents the low-frequency response of my sound system with my subwoofer amp fine-tuned for music reproduction ($f_{CLP} = 48\text{Hz}$). I made all measurements with an SPL meter (Tenmars TM-102) from my listening position, at a moderate sound level. I used C-weighting measurements to perform all adjustments because this type of frequency weighting provides accurate results on the true SPL produced by the equipment.

Looking at this figure, you can see

that the subwoofer improves the frequency response between 60Hz and 40Hz. At 60Hz, the main loudspeaker curve (thin curve) begins to drop, while it remains rather flat when the subwoofer is on (bold curve). Below 40Hz, the contribution of the subwoofer drops sharply due to the limitations of my passive subwoofer. Better extension of the frequency response to the lowest octaves will need to come from a better passive subwoofer.

This graph also reveals weaknesses in my room acoustics. You can clearly observe two dips at 80Hz and 125Hz and also a peak at 40Hz. These irregularities have to do with the dimensions of

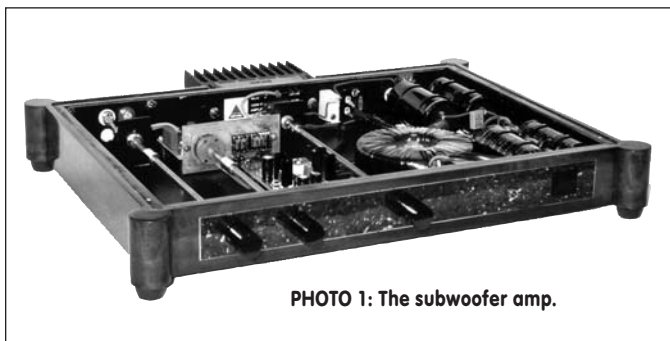


PHOTO 1: The subwoofer amp.

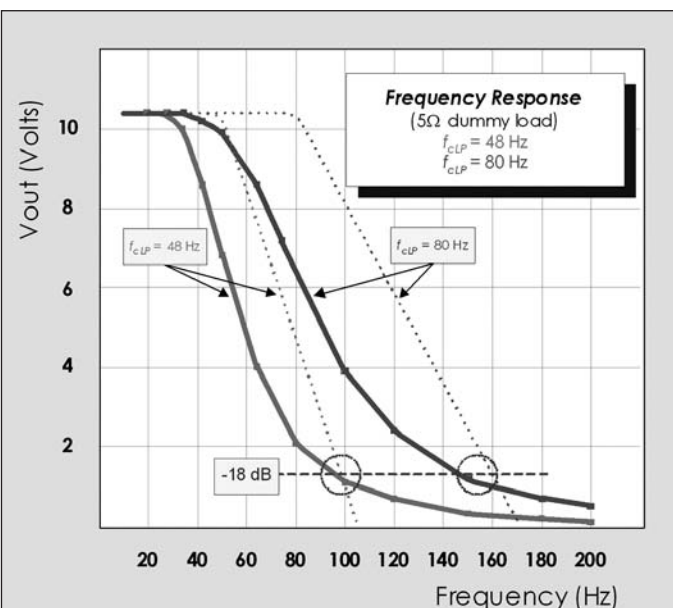


FIGURE 1: Frequency of the subwoofer amp as a function of f_{CLP} .

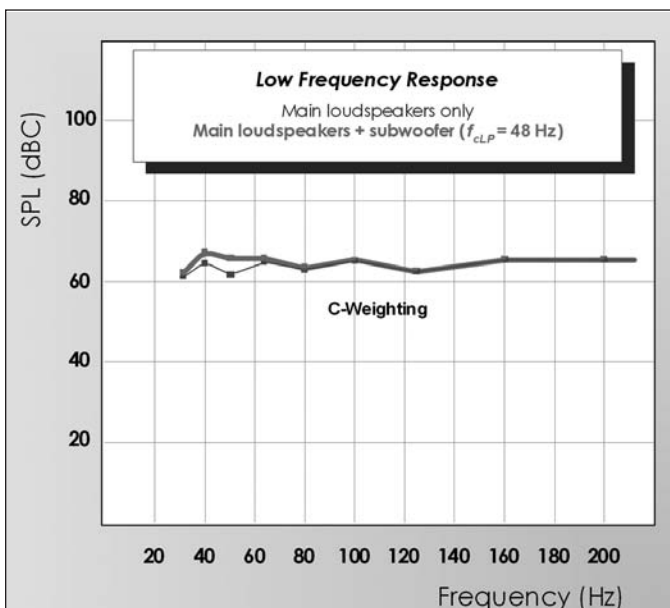


FIGURE 2: Low-frequency response of the sound system with $f_{CLP} = 48\text{ Hz}$.

my listening room. These room modes might be smoothed out with the use of additional wall treatments. Without any doubt, this experiment confirms that the listening room is an important building block of the sound system.

FINE-TUNING THE SUBWOOFER AMP

Integrating a subwoofer to a sound system always offers some challenges to the audiophile. Thanks to the features of the subwoofer amp, it is possible to achieve excellent results. Three parameters need to be optimized in order to provide the best performance: the sound level, the cutoff frequency f_{CLP} and the phase shift.

Adjusting the phase shift is probably the most difficult step of the setup procedure. A good understanding of the behavior of the phase shift is very helpful in making the right decisions. For that reason, I provide some more theory about the phase shift before adjusting the subwoofer amp.

PHASE SHIFT THEORY

In Part 1, I stated that changing the frequency of the musical signal f would change the phase as well. The consequence of this is that the phase can only be optimized for one specific frequency. How will this impact my system? To answer this question, I computed many different values using equation 4 of Part 1. The results are compiled in **Table 1**. An example will show how to use this theoretical data.

Suppose for a moment there is no low-pass filter preceding the all-pass filter stage, so that the musical signal f is not attenuated at any frequency. First play a sine wave test tone of 50Hz simultaneously through the main loudspeakers and the subwoofer. The objective is to get the loudest and cleanest bass possible, indicating that the sound waves of the main loudspeakers and the subwoofer are in phase. Using a SPL meter, set the best position for the phase inverter switch S_3 and then begin to adjust the phase shifter potentiometer P_1/P_2 . After some trial and error, you'll find an optimum adjustment.

Taking a DMM, measure P_1 (or P_2) at, say, 17,750Ω. This determines the cutoff frequency of the all-pass filter

($f_{CAP} = 70\text{Hz}$). Looking at **Table 1**, you find that the phase shift at 50Hz is -142° . Now, suppose you play a signal of 30Hz. The phase shift will be -92° , a change of 50° . Should you play a signal of 100Hz, the phase shift will be -220° ,

a change of -78° with regard to 50Hz.

Are these deviations objectionable? Maybe not, in my opinion. You must remember that the previous stage will be, in fact, a third-order Butterworth low-pass active filter. Its attenuation slope of

P_1/P_2	50 000	47 750	35 000	27 000	21 500	17 750	15 000	12 750	10 900	8 300	4 950	0
f_{CAP}	29 Hz	20 Hz	40 Hz	50 Hz	60 Hz	70 Hz	80 Hz	90 Hz	100 Hz	120 Hz	160 Hz	318 Hz
30 Hz	-184	-180	-148	-124	-106	-92	-82	-74	-66	-56	-42	-22
40 Hz	-216	-212	-180	-156	-134	-120	-106	-96	-88	-74	-56	-28
50 Hz	-240	-236	-206	-180	-160	-142	-128	-116	-106	-90	-70	-36
60 Hz	-258	-254	-226	-202	-180	-162	-148	-136	-124	-106	-82	-42
70 Hz	-270	-266	-242	-218	-198	-180	-166	-152	-140	-122	-94	-50
80 Hz	-280	-278	-254	-232	-212	-196	-180	-166	-154	-136	-106	-56
100 Hz	-296	-292	-274	-254	-236	-220	-206	-192	-180	-160	-128	-70
120 Hz	-306	-304	-286	-270	-254	-240	-226	-212	-200	-180	-148	-82
160 Hz	-318	-318	-304	-290	-278	-266	-254	-242	-232	-212	-180	-106
200 Hz	-328	-326	-314	-304	-294	-282	-274	-264	-254	-236	-206	-128
320 Hz	-340	-338	-332	-324	-318	-310	-304	-298	-290	-278	-254	-180
400 Hz	-344	-342	-338	-332	-326	-320	-314	-310	-304	-294	-272	-206

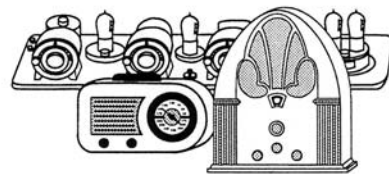
TABLE 1: Phase shifting (in degrees) as a function of the cutoff frequency f_{CAP} and the frequency of the musical signal f .

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HOW TO MAKE TWO SUBWOOFER PREAMPS

Suppose you are interested in the features offered by the preamplifier stage but do not need the amplifier section. What could you do? Adapting the design to make a separate preamp is very easy. Using the same PCBs, I will explain how to build two superb subwoofer preamps.

BATTERY-POWERED SUBWOOFER PREAMP

Imagine you want a simple design. Why not merely replace the power-supply stages—unregulated and regulated—with 9V batteries? Then all you need to do is assemble PCB #1 and 2; add three RCA connectors (one pair of stereo inputs, one output), P_3 , S_4 , a LED, and that's it!

Figure 4 gives you an idea of how you could position the components. Note that I have placed the volume control at the rear of the chassis. This configuration minimizes the wire length and also keeps the unit as small as possible. In this example, a 5" × 7" × 2" Hammond chassis would fit nicely.

You could upgrade this preamplifier by replacing the 9V batteries with a regulated wall-mount transformer. Models such as the WM063 from Elpac Power Systems provide $\pm 12V$ DC and 5V DC connections with an output ripple of 20mVp-p. The $\pm 12V$ DC should feed PCB #3 to further improve filtering, while you could utilize the 5V DC to power the LED.

The proposed layout offers sufficient room to accommodate the placement of PCB #3 in lieu of the 9V batteries. Figure 5 also suggests a flexible approach that will ease the upgrade of a battery configuration to a regulated wall-mount configuration, with the simple addition of a 3PDT switch.

FULL-FEATURED SUBWOOFER PREAMP

You may prefer a preamp that would somehow replicate the design of the subwoofer amp without its amplifier module. To keep the preamp small, I recommend the use of a wall-mount transformer for this project. Figure 6 suggests that you could build such equipment using a 5" ×

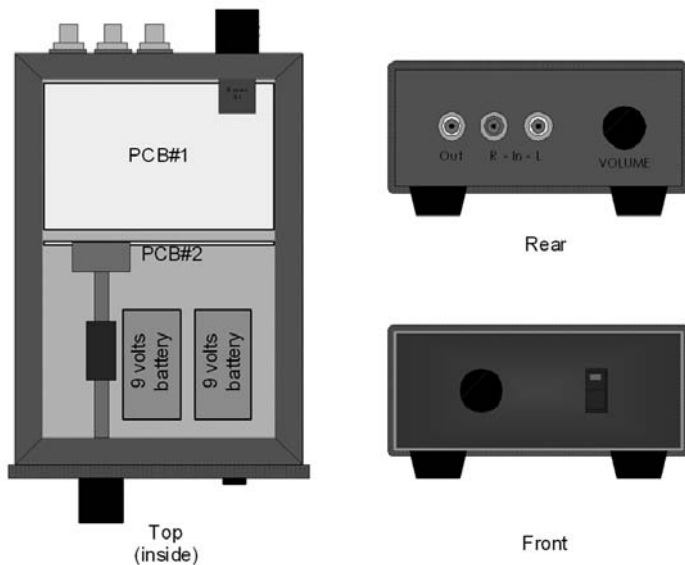


FIGURE 4: Sketches of the battery-powered subwoofer preamp.

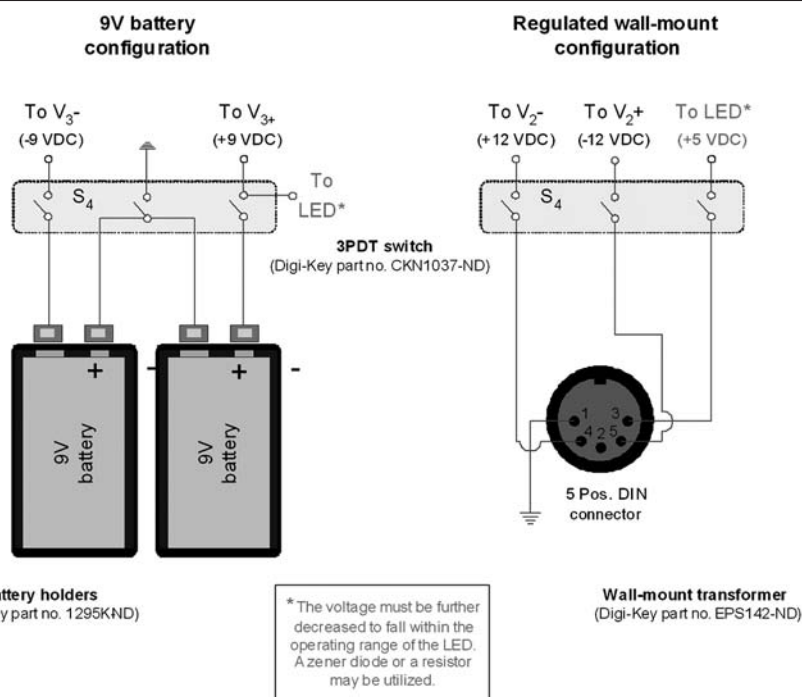


FIGURE 5: Suggested connection of S_4 for the subwoofer preamps.

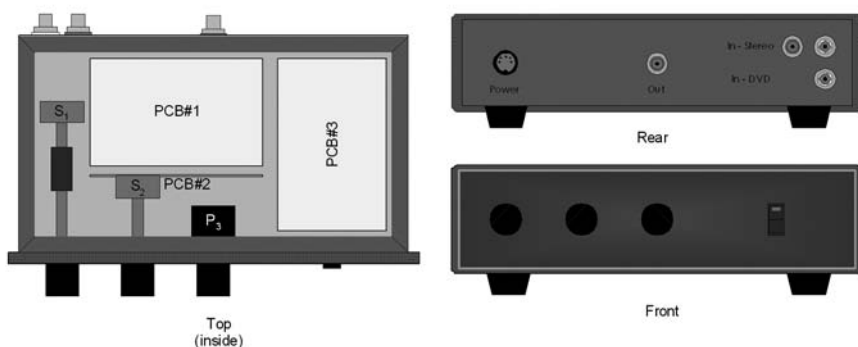


FIGURE 6: Sketches of the full-featured subwoofer preamp.

-18dB per octave will greatly attenuate the effects of these unwanted deviations as the frequency of the signal increases.

The key here is that the cutoff frequency f_{CLP} of the third-order Butterworth low-pass filter should be in the vicinity of the cutoff frequency of the main loudspeakers. The lower the crossover point, the better. On the other side of that cutoff frequency, a good subwoofer will take over the main loudspeakers down to the lowest octaves. So whether there is a deviation at these frequencies should not seriously impact the end result.

All this is a simplification of the reality, however. To be truly accurate, you would need to consider the phase shifts introduced by all other filters in the circuit, and mainly by the third-order Butterworth low-pass active filter. It is important to understand that the phase

shift will be optimized not only for one frequency f but also for one specific cutoff frequency f_{CLP} .

On the basis of that knowledge, I chose to adjust the phase shift to get the highest fidelity possible when listening to music, and aligned the phases while using the cutoff frequency f_{CLP} dedicated for music listening. You may wish to set a different alignment. In any case, **Table 1** will help you predict the behavior of your setup.

ADJUSTING THE VOLUME LEVEL

I connected the stereo outputs of my preamp to both my dual mono tube amps and my subwoofer amp. This configuration allows me to control the sound level of the entire system with my preamplifier.

As Fletcher and Munson demonstrated a few decades ago, our ears do not perceive all frequencies with the same ability. So it is important to make sure that you perform all your adjustments on the right basis. I suggest that you first listen to music at the volume level you are used to. Then using an SPL meter, try to figure out the average SPL of the

musical content. This will be your starting point. In my case, I averaged an SPL of about 65dBC.

From there, you need to set the volume of the main loudspeakers and the subwoofer, separately, to this level. Using a test disk (*Stereophile* Test CD 3, track 17), I played a test tone of 60Hz and adjusted the volume levels to get about 65dBC. The value of 60Hz is arbitrary.

What is important is that you must use a frequency higher than the cutoff frequency of your main loudspeakers (40Hz in my case) but lower than the cutoff frequency of the subwoofer amp. I used the cutoff frequency f_{CLP} of 100Hz to make sure the 60Hz signal was not attenuated when I adjusted the volume control. If you need to use a higher frequency to perform this procedure, I suggest you prepare a small chip of resistors that will provide you with a higher cutoff frequency.

CUTOFF FREQUENCY AND PHASE SHIFT SELECTIONS

Your objective now is to find the optimum cutoff frequency f_{CLP} and phase alignment for the subwoofer amp. Ideally, the sub-

9" x 2" Hammond chassis.

You could also try a 15 VA toroidal transformer providing dual secondary windings of 12V AC. In this case, PCB #4 rectifies the AC voltages and provides approximately $\pm 16V$ DC, just perfect to power PCB #3. You would need to modify some components (referring to Fig. 10 of Part 1).

Replace the 4700 μ F/80V capacitors, skip ZD₁ to ZD₅, and replace them with jumper wires. Replace ZD₆ with a 14V/5W zener diode. D₁ to D₄ would remain the same, or, you could replace them with a lower voltage type. As for R₂₆/R₂₇, you could simply replace these resistors with jumper wires because the inrush current will be limited. The only downside of this configuration is that you would need a larger chassis to receive PCB #4 and the power transformer.

Whether it is a battery-powered or a full-featured subwoofer preamp you intend to build, I recommend you use only shielded interconnect cables between the preamp and all other components. The design seems to be vulnerable to RFI, and you should do everything to minimize that susceptibility. Aside from this, I am truly confident that either preamp works just fine. ■

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woofer and the main loudspeakers should blend in a seamless manner while extending and flattening the frequency response to the lowest octaves.

Experimentation was very easy through the use of five chips of resistors that provided me with the following cutoff frequencies: 35Hz, 39Hz, 44Hz, 48Hz, and 52Hz. Within seconds, it was possible to swap one chip for another.

I made the first set of measurements with my main loudspeakers alone. I played test track 17, from 200Hz down to 31.5Hz (the lower limit of my SPL meter), and computed the SPL values in a spreadsheet from which I produced the thin curve in **Fig. 2**. The frequency response then appeared quite bumpy, revealing room acoustic problems as mentioned before.

I carried out the following procedure for each chip of resistors:

1. Using a test tone of 40Hz, the cutoff frequency of my main loudspeakers, I played that signal simultaneously through the main loudspeakers and the subwoofer.

2. I first switched the phase inverter S_3 in the position providing the loudest bass. (Note: Once you do this, you do not need to modify this setup for the subsequent

chips.)

3. Then I made various attempts to adjust P_1/P_2 and found an optimum adjustment after a while. SPL measurements indicated that the bass was at its loudest (67dBC).

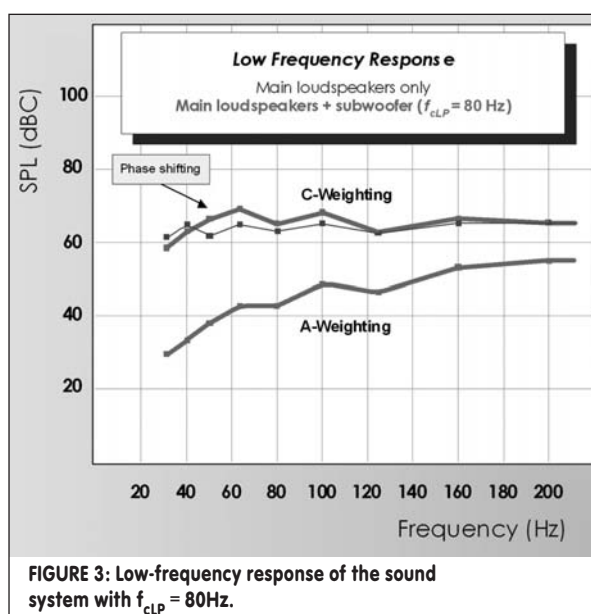
4. I turned down the volume of my subwoofer amp to get an overall level of 65dBC.

5. Finally, I played test track 17 and computed the values in my spreadsheet. You might need to redo this step a few times because the volume level might require further correction to get the flattest response possible.

After completing the procedure for all five chips, I compared the data and concluded that the 48Hz cutoff frequency was the optimum choice for music listening.

FINAL THOUGHTS

After completely fine-tuning my sub-



woofer, I decided to make some final SPL measurements, but this time with my preferred cutoff frequency for home theater ($f_{CLP} = 80$ Hz). The data computed in **Fig. 3** led to some very interesting observations. The subwoofer emphasizes the frequencies from 125Hz down to 50Hz. The third-order Butterworth low-pass active filter does a pretty good job, because the

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influence of the subwoofer disappears over 125Hz. However, below 50Hz, you can see that a phase shift occurs as the SPL becomes lower with the subwoofer compared to the main loudspeakers alone!

This experiment clearly demonstrates that the phase alignment can only be optimized for one specific cutoff frequency. Maintaining perfect alignment for each of the three cutoff frequencies would require you adjust P_1/P_2 every time. Because I had no intention of fiddling with this potentiometer, I positioned it inside the amp, directly on PCB #1. Music being my priority, I can easily live with this deviation when I listen to a movie. Should your preference be different, you will need to experiment further to determine a different compromise.

Out of curiosity, I also measured the SPL using the A-weighting that is meant to correlate to the behavior of the human ear. My physiological perception was somewhere between the C-weighting and the A-weighting, but the tendency confirmed the works of Fletcher and Munson. Looking at this curve in **Fig. 3**, you could argue that our ears are not electronic instruments and that subjectivity should be allowed when setting the volume level. To a certain extent this is right.

If you tuned your system on the basis of a moderate SPL (65dBC in my case), you may wish to readjust the volume level of the subwoofer amp a little higher so that you have a better physiological perception of the lower spectrum. You must be careful, however, not to exaggerate. You should never feel the presence of the subwoofer in the room. You should extend the low-frequency content to the lowest octaves and thus bring some support to the main speakers. Also, instruments should sound in your listening room as close as possible to the way they would sound in a concert hall. Having said that, music is a matter of preference and I am sure you will tune it to your taste!

One last word about tuning. I have not talked about subwoofer placement. This is another important aspect I intentionally omitted because in my room my passive subwoofer can be positioned in only one place. If your room offers you the flexibility of moving your passive subwoofer around, then I encourage you to experiment with its placement.

LISTENING IMPRESSIONS

It is well known—and I can confirm this—that the use of a subwoofer adds to the movie experience. Aside from the spectacular sensations provided by the low-frequency effects, I found that the clear winner is ambiance, which offers me the greatest satisfaction. The feeling is subtle but very important; it really gets you into the movie.

What is true for movies is also true for music. Adding a passive subwoofer controlled by a full-featured subwoofer amp contributed to my listening experience. Instruments such as piano, organ, and double bass are more involving and emotional. For example, when the pianist hits a low bass note on a grand piano, I feel more intensely—almost physically—the full weight of that note. The play is more realistic, closer to the real thing. And because the cutoff frequency is very low, I never perceive any adverse effect of phase deviations between the subwoofer and the main loudspeakers.

Again, the adjustment of the volume level is highly important. Whether you listen to a movie or to music, you should

never hear the subwoofer itself.

CONCLUSION

I hope you have enjoyed the description of my journey involving the design and construction of my subwoofer amp. And, more important, I hope it has inspired you in your own projects.

Should you attempt to build this subwoofer amp or one of the preamps, I will be glad to offer you some support. Readers can communicate with me at my e-mail address r.godmaire@sympatico.ca. I will do my very best to answer e-mails in a timely manner. Enjoy the hobby! **ax**

Note: We are pleased to announce that the four PCBs (on a single board) are now available through *audioXpress* as a kit for \$60.

A word from the author: Since the submission of my article, I have performed several improvements to my boards which you will see in this kit.—R.G.

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DeNoising a Vanilla Preamp

By Darcy Staggs

A manila folder dwelt for a long time in my file cabinet with a hastily scribbled, unimaginative title on the tab—"Preamp X." I had started a preamp project nearly 20 years ago and collected data for that end, but ultimately let the whole affair lie dormant, as I realized that my design ideas weren't "high end."

Not so long ago a friend sent me some suggested circuits and parts recommendations which looked very inviting, so after I scrapped old boards and unused documentation and resurrected a half-made chassis, Preamp X was finally born (Photos 1 and 2).

This article is not a preamp construction project at all, but rather documents what happened to Preamp X once it began to reproduce sound. I offer excerpts from my log of Preamp X to illustrate the difficulty in properly finishing even a simple piece of electronics in a way that brings out the best of its realistic musical potential.

BASIC FEATURES

Preamp X houses two PC boards mounted on edge. One board contains two op amp phono stages (with inductor RIAA rolloff), while the other contains a pair of op amp line drivers. Separate zener regulators provide power at each op amp supply pin. The raw supply is located behind a bulkhead to successfully reduce the radiation of noise into the circuitry.

There is a five-way input selector switch, a blue Alps volume control, and similar balance control. Toggle switches provide "muting" and "mono" functions. The preamp was built into a Sunix chassis made of 0.035" steel (six elemental panels) with an aluminum front dress

panel. As you can see, in no way is Preamp X electronically or mechanically unusual, although component selections followed the designer's exhaustive testing.

FIRST VOYAGE

After the customary startup debugging and tweaking, the preamp came alive and began turning out "promising" sound, meaning I didn't think the promise was fulfilled. Certain aspects of the music were acceptable, but the mainly bland presentation was far from "live." So I began a carefully documented witch hunt to determine whether vibration was a factor. (The preamp has always rested on four Sorbothane feet, a practice I observe for all electronic components.)

THE LOG

I have separated out the mechanical change items from the current total of 32 entries in the project logbook for the preamp, and repeat them here along with their noted sonic consequences. Where I underlined my logged comments, upper case is used here. Similar entries accompany the electrical alterations, but those are not the subject of this article.

In many cases I made several

Follow this author's vibration-reduction tips to achieve some rewarding audio results.

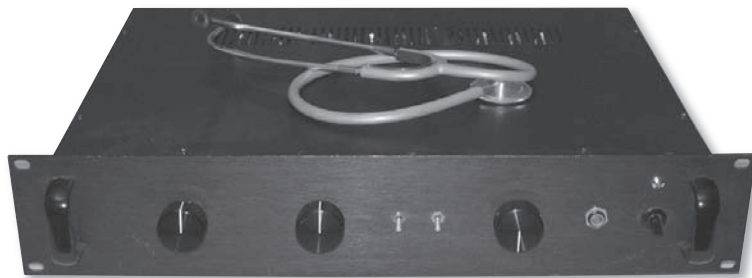


PHOTO 1: Preamp X (with medical stethoscope to detect troublesome noises).

physical changes at once, clouding the issue of which produced the most sonic improvement. If I build another preamp, I would complete all items listed during the initial construction phase, because they all tested successfully. The basic mechanisms I attacked were panel and component resonance and colliding objects:

2. Coat op amps and raw supply caps with lead foil (two layers). Note gains in detail, bass, linearity, clarity.
3. Coat all mating cabinet surfaces with lead foil. Put long strips on top, bottom, side panels. Improvements similar to 2 above.
9. Mount PC board L-brackets to grommets holes instead of bare chassis metal. Small improvement in clarity.
12. Mount front-to-back 1/4" x 1/2" oak strips on top, bottom covers (two each) with Gorilla Glue (urethane based).



PHOTO 2: Preamp X insides.

Add lead foil to power supply bulkhead. Apply remaining scraps of self-adhesive damping "tar paper." **LARGE** gains in detail, clarity, overtones, musicality.

15. While open (to replace phono co-ax cable), add more lead strips, some foam, and rubber pads to combat vibration between the front dress panel and the steel structural panel immediately behind it. Add foam dampers to the long wires running between the input jacks and selector switch. **MAJOR IMPROVEMENT** in sound quality—sense of liveness, purity, dynamics.

It was startling to realize that items 12 and 15 were very successful **AFTER** adding rubber grommets to the PC board mounts. However, the entire raw supply, controls, and jacks are still hard-mounted except for the foam pad supplied with the toroidal transformer, so evidently something there is vulnerable. Because the results are in hand, I won't spend time isolating the culprit(s), but some mystery lingers here.

DOWNSTREAM

The task of Preamp X is to drive a sec-

ond electromechanical contraption, "Amptron," another collaboration using the designer's electronic specifications and my chassis.

Here I figured I had applied my structural lessons from Preamp X, but found that Amptron still needed to undergo the same witch hunt, and that it too reacted strongly to even more sophisticated vibration control than appears here (the subject of another article).

FINDINGS

The inescapable evidence from these fruitful experiments is that electrons should be the only moving parts in an electronic system. This can be troublesome to achieve depending on how loud the music plays, as I learned while surveying the structures with a medical stethoscope, a very revealing instrument.

The sum total of these modest efforts resulted in a remarkable unveiling of the musical presentation. Actually, I'd rather be telling you that the changes are small and this effort was needless, but the fact is, Preamp X has truly blossomed forth. *ax*

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Build This Six-Transistor Low-Noise MC Preamp

By Rickard Berglund

This six-transistor low-noise distortion MC step-up uses a new circuit topology with three feedback loops. Input signal is fed to the emitter of transistors Q1 and Q4 and is amplified by transistors Q2 and Q5. Transistors Q3 and Q6 work as an output buffer. The collector voltage of Q1 and Q2 acts as the operating current of Q2 and Q5, and the resistors R4-7 set the operating voltage of transistors Q3 and Q6. With an excellent drive capability and stability into capacitive loads, the amplifier is symmetrical and is DC-coupled from input to output. Input impedance is 120Ω; output impedance is 200Ω. The bandwidth of the amplifier without L and C1 is 0-6MHz.

The output offset voltage is typically less than 10mV. If the offset voltage is a problem, place a coupling capacitor between the MC step-up and the RIAA

amplifier. The gain is set by the resistors R4 and R5 (Table 1). The inductance L and the capacitor C1 act as a filter to suppress radio frequency interference.

TABLE 1.

R4, R5	Gain	Gain in dB
240Ω	12	21.5
270Ω	14	23
300Ω	16	24.5
330Ω	20	26

You can power the amplifier from a 12V lead-acid battery or a low-noise 12-18V power supply. Power consumption is 27mA with a 12V supply (325mW), 33mA with a 15V supply (500mW), and 40mA with an 18V supply (720mW). Shield the amplifier in a metal box for shielding, and place transistors Q1 and Q2 together so they have the same temperature. Also place transistors Q4 and Q5 close to each other. Eventually, glue the transistors to each other. *ax*

PARTS LIST.

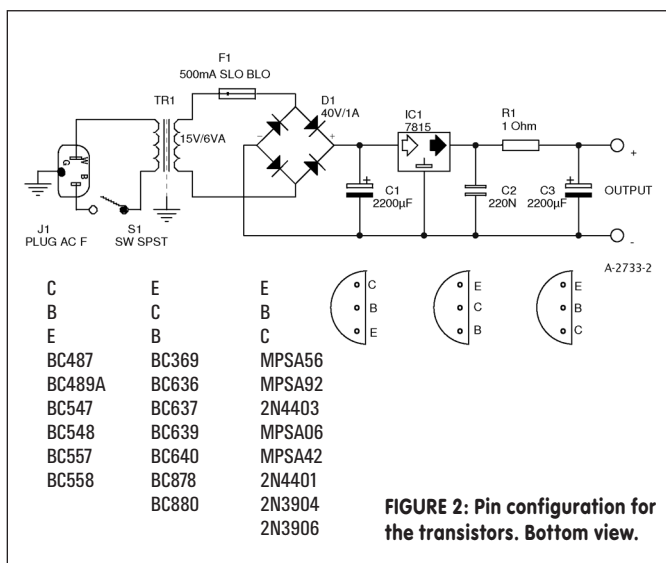
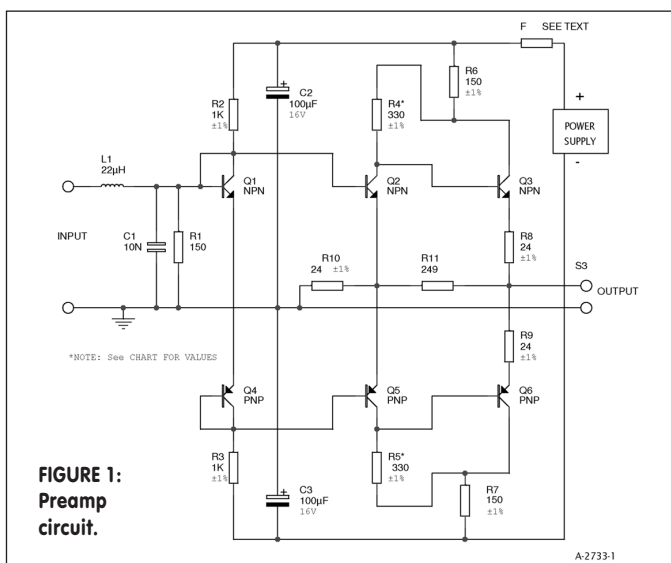
C1	10nF
C2, C3	100μF 16V electrolytic
F	fusc 100mA
L	22μH
Q1, Q2	low noise NPN transistor
Q3	BC547B or BC548B or 2N3904
Q4, Q5	low noise PNP transistor
Q6	BC557B or BC558B or 2N3906
R1	150Ω
R2, R3	1000Ω 1%
R4, R5	240-330Ω 1% (see text)
R6, R7	150Ω 1%
R8-10	24Ω 1%
R11	249Ω

Transistors Q1 and Q2	Relative Noise
BC 639 Motorola	66
BC 489A Motorola	71
MPSA 42 ON Semiconductor	73
MPSA 42 Motorola	73
BC 639 Fairchild	75
BC 637 Siemens	77
BC 487 Motorola	82
BC 635 Philips	83
2N4401 ON Semiconductor	87
MPSA 06 Fairchild	114
MPSA 42 Fairchild	114

Transistors Q4 and Q5	Relative Noise
BC 878 Siemens	32
BC 880 Siemens	33
BC 636 Motorola	58
BC 640 Motorola	65
MPSA 56 National	66
MPSA 92 Motorola	67
MPSA 92 Multicomp.	74
BC 640 Multicomp.	74
BC 636 Siemens	75
BC 640 Fairchild	77
BC 369 Philips	98
MPSA 56 Fairchild	100
2N4403 Motorola	101
BC 640 Philips	114

PARTS LIST FOR A LOW-NOISE POWER SUPPLY (ONE POWER SUPPLY FOR TWO MC STEP-UP AMPLIFIERS IN A STEREO SYSTEM).

Br	bridge rectifier 40V 1A
C1, C3	2200μF 40V electrolytic
C2	220nF
F	500mA slow-blow fuse
IC	7815
R	1Ω 1W
T	transformer with a 15V 6VA secondary and a shield between primary and secondary





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Mouser Electronics released its Catalog No. 628, featuring 1,848 of the newest products and latest technologies supplied by the company. This catalog contains 63 additional pages of new product information and new section guides, as well as colored banners and logos indicating new products, technology, and suppliers. In other Mouser news, the company signed a distribution agreement with **THAT Corporation**, adding THAT's balanced line drivers and receivers, microphone preamplifier ICs, and voltage controlled amplifiers to its inventory. For more information, go to www.mouser.com.



PS Audio debuted the Trio C-100 Control Amplifier, based on Gain Cell technology, producing 100W into 8Ω and 200W into 4Ω. Other versions are the P-200 Stereo Pre-amplifier, which has the ability to drive any power amp; and the A-100 Power Amplifier, which has a high input impedance, making it an ideal load for vacuum tube amps. PS Audio also released the Power Plant Premier, which regenerates perfect AC power; the Quintessence Power Center, incorporating leading-edge Nano-Crystalline filters essential for performance while providing increased AC noise purification; the Quintet and Duet Power Centers; and the Soloist In-Wall Power Conditioner. For more information, call 1-715-425-8826, or e-mail matt@psaudio.com.



AVX Corporation has released its "What's New at AVX" catalog, highlighting over 18,000 new parts and 60 new product families introduced in the past year. It contains 62 product families and details on new software and literature, including new capacitors, timing devices, connectors, filters, and more. To learn more, visit www.avx.com.



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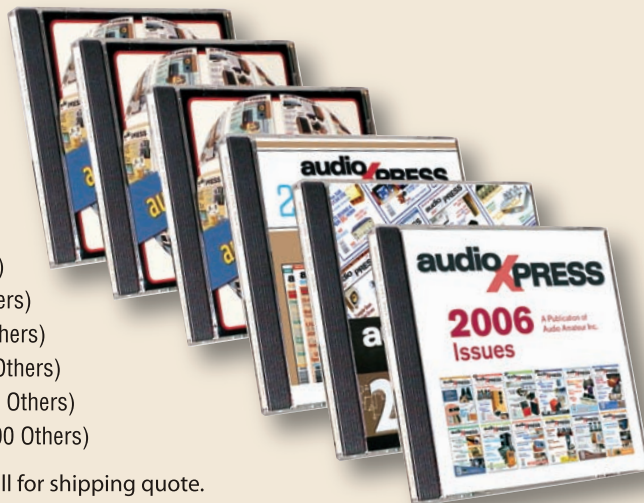
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SPEAKER MODS

Nice article by Ed Simon ("A Combination Horn You Can Build," Jan. '07, p. 10). His description of the bass properties of the horn sounds remarkable.

Is it possible to eliminate the midrange-supporting section of the speaker and optimize the horn for reproduction of bass frequencies only? What would that look like?

Since the woofer would not be used for midrange frequencies in this application, how about configuring it so that the woofer fires inward, and the output of the horn is facing the listener instead of facing into the corner?

Angel Rivera

angelluisrivera@netzero.com

Ed Simon responds:

Yes, you can use this as a bass only speaker. Do not cut out for the woofer in the front of the box. Mount the woofer in the filter board that forms the top of the Z section. Extend this board 2" in length to connect to the front of the box. The rear chamber should now be sealed.

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You can now decide whether you want to make a bass reflex loading on the rear chamber of the woofer by putting a port tube in the front. I would do this near the bottom of the chamber on the face of the loudspeaker. If you are placing a small speaker on top of the woofer, you may wish to place the port high near the full-range unit.

If you want to try this inexpensively, make the test enclosure out of several pieces of corrugated cardboard glued up to about 1" thickness. Any old 8" speaker should work. The lighter and stiffer the cone, the better for this application. I would not use a speaker designed to be a small box woofer because they tend to have a lower output and the heavier cone does not match as well to the horn load.

If you try this, let us know how it goes.

LPS TO DVDS

First, I would like to compliment Victor Staggs on his excellent article, which was both thorough and informative ("Transferring LPs to DVDs in High Resolution," Feb. '07, p. 14).

The reason I'm writing is because I'm also planning a similar project, and I have some additional information that may be of use. (Note that I am planning to do this on a PC.)

I would strongly suggest that unless you plan to follow Victor's excellent instructions exactly—using similar hardware and software—you become familiar with the process of producing and recording DVDs in general before making decisions about DVD formats.

Converting analog to digital. The entire part of the process involved in converting LPs into a hi-res digital format is the same regardless of your final choice of medium. The digital files will be immediately playable on a computer—or may be output to an outboard DAC. You will need to be sure that your computer card and DAC have compatible connections—TOSLINK or coax.

You should also be aware that DACs vary in the formats they support; many older DACs may not support 96/24 digital or higher, and some newer ones may still not support 192/24. You can do this part before even deciding what DVD format to use.

DVD audio formats. There are two entirely different choices available for putting audio on a DVD: DVD-Audio and DVD. The method that Victor uses is considered DVD (he is producing a standard video DVD with hi-res PCM audio and blank video). You should be able to play Victor's method on most DVD players—although some may limit their digital output format in terms of resolution (some players simply will not output 192/24 audio from their digital ports—and some may not be willing to do 96/24).

There may even be players out there that do not support playing 96/24 PCM audio from a DVD—not all players support all formats in the standard. The main problem here is that this is often simply not specified in the player's spec sheet. DVD players are cheap—this just means that you might need to search for one that does what you need.

You may also have a problem with software—a lot of DVD recording software (especially low-cost programs and shareware for the PC) simply fails to specify what formats it supports. They say "put audio here"—you drop it in and see whether the software accepts it or not. Worse, some software will automatically down-sample the audio without telling you. The settings may be somewhere in the options or you may need to call product support for an answer.

I suggest making sure that the software you choose accepts using hi-res audio before experimenting (and that it will not generate an error if the video stream is blank). The software that Victor used actually produces the DVD content and outputs it as a video file—as I understand it.

Depending on the software you choose, some PC software will allow you to separately specify audio and video content—in which case you may be able to simply not specify a video file (the step of producing the combined audio/video file is carried out internally).

The other alternative choice is to use DVD-Audio, which is a different standard and somewhat more difficult. To play it you specifically need a DVD-Audio compatible player. There is a DVD-Audio copy-protection scheme that limits the digital output resolution of tagged commercial disks (or blocks it entirely); this is

not an issue on disks you record. In addition, however, many players do not play the bits from the DVD straight out the digital output (this is called “bit transparent” when it is mentioned). Another issue is that there is very little software that supports recording in true DVD-Audio format—and most of it is expensive.

One benefit to DVD-Audio is that it supports a type of lossless compression that can double the capacity of a disk—the downside is that the codec is licensed by one company and costs about \$2500. Odds are that Victor’s choice is easier to implement and more universally playable. The quality should be similar, but there are “audiophile quality DVD-Audio players” out there that may be better than combined DVD units.

DVD players. There are just a few facts about DVD players in general that you should know:

- some DVD players will not play DVD-recordable media (the ones that do cost no more than those which do not, just be sure you have one that does)
- DVD-recordable media is divided into DVD-R (“dash R” or “minus R”) and DVD+R (“plus R”); they are similar and most computer drives support both, but your recorder and player must support the one you choose or it won’t work (some players support only one or the other)
- some DVD players do not support hi-res digital output or even have a digital output (be sure yours does if you plan to use an outboard DAC)
- most DVD players have a coax (S/PDIF) digital output and/or an optical (TOSLINK) output; some DACs have only one or the other input; make sure both of yours support a common format if you plan to use an outboard DAC
- many DVD player TOSLINK outputs do not support 192/24 digital format (apparently much of the standard TOSLINK hardware tops out at 96/24); on some of these machines the S/PDIF output is also limited to 96/24; on some it supports 192/24

I hate to make it seem so complicated, but this is one of those subjects that becomes very convoluted once you “get off the path.”

One final note: Some DVD software has the option of producing a disk that is immediately playable. This usually involves a menu option and should result in a disk

that plays as soon as you put it into the drive—so there is no need to push any buttons (or remember what to push without a screen connected).

Keith Levkoff
kLevkoff@panix.com

TRANSISTOR NOISE

Some questions are raised in the February issue of *aX* about transistor noise (“Xpress Mail,” p. 60). If the letter writers will send me—via *aX*—a few of each type of their transistors in question, I will measure them and report back on the results via *aX*. I use a Quan-Tech Labs Transistor Noise Analyzer.

I have found the noise specs to be wildly different in some cases from the spec sheets. Some with high noise specs are very quiet and vice versa. It appears that as factory processes “improve” over the years, the effects/changes on the spec sheets don’t change from the original. This seems more likely with transistor numbers that have been around for a long time.

The bottom line is that they need to be checked/tested. Then you find out the noise with varying DC currents and voltage to find optimum noise value conditions.

Richard Marsh
Cool, Calif.

CF FOLLOW-UP

I wish to thank Christopher Paul for his in-depth analysis of the relative performance of the cathode through mu follower to SRPP (“The Cathode Follower and Its Weaker Siblings,” Jan. ’07, p. 22). The final comment made in the article disturbs me somewhat, because, although the cathode follower (CF) may demonstrate the “gold standard” for the measurements undertaken, we have no basis to assume that it may in any way sound superior.

An FFT harmonic distribution profile—either as measured or from a Spice simulation—would point the way more clearly as to the advisability, or not, of discarding the SRPP.

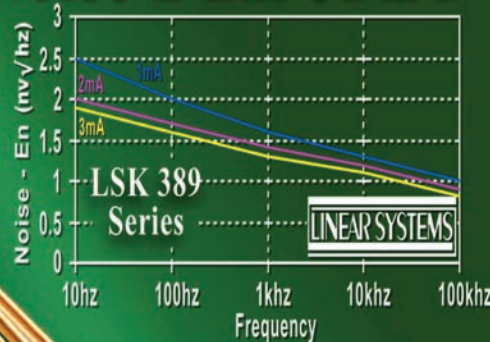
Wayne L. Wright
New Zealand

Christopher Paul responds:

I thank Mr. Wright for his interest in my article and acknowledge his concerns about how the circuits sound. I do, however, question his certitude that it offered no guidance at all on this topic. Surely it’s clear which stage would suffer the most sonically when powered from a less than clean supply or driving a potentially bandwidth-limiting capacitive load.

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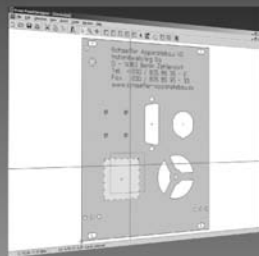
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I have generally steered clear of addressing questions of what sounds superior in my articles. I believe that this topic is complex and requires more work to address authoritatively than I care to attempt. Mr. Wright's assertion that there is a correlation between an FFT spectrum and sonic quality is debatable, unless we're discussing gross distortion. There is much literature supporting a number of different views on this topic. Which should we adopt?

Suppose that we were to accept a specific correlation, no mean assumption. To generalize about the relative distortions, it would be necessary to test a wide variety of valves as SRPPs, CFs, and MFs over a range of bias conditions while driving a number of different loads, capacitive and not. And you would need to consider the balance of distortion characteristics from tube to tube in the pair of SRPP triodes, because that configuration tends to subtract its triodes' even (2nd order) distortion as it averages its (3rd order) odds. As I said, a lot of work.

What may be instructive, however, is to simulate the AC grid-to-cathode voltages of my article's SRPP and CF at common output levels. I think it's fair to say that these voltage levels will indicate the relative amounts of distortion that each triode is generating. I'll assign $\mu = 30$ and $gm = 0.012$ mhos (siemens). With equal output voltages across a load of 100K, I get SRPPbottom/CF = 1.07 and SRPPtop/CF = 0.74. For a 10K load, I get SRPPbottom/CF = 0.93 and SRPPtop/CF = 0.75.

The first thing you see from these results is that there are no significant differences in the drive levels of the three triodes. The second thing is that the average of the SRPP levels is slightly less than the drive of the CF. Because the SRPP averages the odds but subtracts the evens, I'd expect from the SRPP less 2nd order (the total amount depends on the balance in the distortion characteristics of the two triodes) and only slightly less 3rd order harmonics. You can be the judge as to whether these small differences mean anything.

In summary, considering the small differences expected in distortion and the big differences in Gain, Zout, and PSRR, I'll stick by my article's limited sonic implications listed here, and es-

pecially to the conclusion that if you choose to use the SRPP, you'll now know what you're losing.

TRANSFORMER MAGIC

I think Bob McIntyre did a great job on his 25L6 "Mighty Mouse" amplifier, which appeared in April '06 (p. 34). I'm amazed that Bob got great results using power transformers as audio output transformers. In spite of the great sound, I'm sure that there are those who don't feel comfortable substituting audio output transformers with power transformers, and have shied away from building the amp. If you wanted to build the "Mighty Mouse," and use audio output transformers, what impedance would you use?

Because the power transformers Bob uses are 240V AC CT:12V as connected, the primary-secondary ratio is 20:1. If squared, and multiplied by the load impedance connected to the secondary, 8 Ω , as is the formula for determining primary impedance, the result is $20^2 \times 8$, or 3200 Ω C.T. In looking through the Antique Electronic Supply catalog, there is one such transformer available as a replacement in Marshall guitar amplifiers, but it's rather large, maybe too much so, for the "Mighty Mouse" chassis.

A second option is to use the Hammond P-T1609 (10,000 Ω CT:4-8-16 Ω) and connect the UL leads to the plates instead of the blue and brown wires which should have their ends taped, or soldered to an empty terminal lug. The red wire would be connected to B1 (150V DC) as usual. Also, the 8 Ω speaker would be connected to the P-T1609's 4 Ω tap, instead of the 8 Ω tap. This configuration provides the 20:1 ratio for 3200 Ω C.T.:8 Ω .

The P-T1609, like all Hammond 1600 series output transformers, has its UL taps at 40% of the primary. To determine the UL-to-UL impedance, multiply the primary value by 0.16. If, for example, the primary impedance is 10,000 Ω CT, the UL-to-UL impedance is 1600 Ω CT, not 4000 Ω (40% of 10,000), as you might believe. To understand why this is, you need to diagram a 10,000 Ω CT:4-8-16 transformer with 40% UL taps (located toward the center tap) as I have done in **Fig. 1**, based on the necessary turns ratio.

The ratio of the transformer is found using the formula



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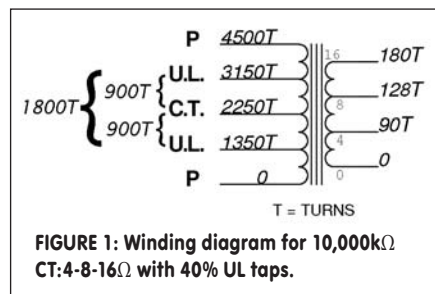
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$$\sqrt{\frac{Z_{pri}}{Z_{sec}}} = \sqrt{\frac{10000}{16}} = 25(1)$$

The 25:1 ratio tells you that the primary will contain 25 times as many turns as the secondary. The example in **Fig. 1** has 180, therefore the primary has 4500. The center



tap is located at 2250 turns, and the UL taps at 1350 and 3150. The UL winding, 40%, consists of 1800 turns, 900 turns each side of the center tap.

The secondary, which again has 180 turns, is tapped at two locations. Because the 4Ω tap is located in the middle of the 16Ω secondary, it will be found at the 90 turns point.

To find the 8Ω location, you must find the ratio of 10,000Ω:8Ω by using the earlier mentioned formula

$$\sqrt{\frac{10000}{8}} = 35.36$$

Next, divide the number of primary turns by this result

$$4500 \div 35.36 = 127.26 (128)$$

The 8Ω point is at 128 turns—not 135 turns, the halfway point between the 4Ω and 16Ω taps, as some might assume.

Now knowing that the diagram in **Fig. 1** is correct, you can determine the UL-to-UL value. Simply divide the number of turns in the UL winding, 1800, by the 180 turns in the secondary, to get 10, or 10:1. To find the UL-to-UL impedance, square this result and multiply it by 16

$$10^2 \times 16 = 100 \times 16 = 1600\Omega \text{ CT.}$$

The UL-to-UL value is 1600Ω CT, or 16% of the primary value. Because all Hammond 1600 series transformers have 40% UL taps, you can determine their UL-to-UL value by multiplying the primary impedance by 0.16 as well.

By knowing their UL-to-UL impedance, you can use Hammond 1600 series transformers in many more applications.

Neal A. Haight
Castro Valley, CA

LISTENING TESTS

Why doesn't someone write an article for readers who don't possess a lot of bench test equipment (or no test equipment at all)? Then, other readers could suggest recorded source material—not just special test CDs—to use for a really comprehensive listening test to “flex the muscle” of their system?

I have no test equipment; however, there are certain recordings I always go back to so that I can tell whether I picked up a quality component or, whether the ones I've rebuilt are at their optimal performance.

As much as I feel like a kindred spirit to Dennis Colin and Charles Hansen, the “Rube Goldberg”-type technical minutiae doesn't communicate the “worthwhile immediacy” any audiophile felt (or not) if they'd been sitting right in front of those Scott tube amps. The principle I go by is that an audio component—regardless of price—will never match the sound of live music (even, probably with a \$37,000 Audio Note Ongaku monoblock amp). That's not its job. But what it should do is extract every ounce of signal from the source it can in the most distortionless and transparent way.

I prefer to do my listening tests with old, analog reel-to-reel tapes. (I was handed down a library of them by my father—all still in good condition, too.) I believe (be it my hangup or be it true) that these represent the closest thing to a “time travel back to” vintage studio performance (because they don't amass the kind of surface degradation vinyl records are susceptible to).

The deck I use is a (rebuild) AKAI 1730-SS, two speed (7½ and 3.75 ips) quadrasonic recorder from 1973. (It originally sold for \$349.) I've replaced the following components:

- all the capacitors on the playback and record preamplifiers as well as on the power supply
- the TO-220-type voltage regulator with a NTE-brand substitute
- the power cord with 12AWG wire and a heavy duty Hubbell plug (this had the most dramatic effect on the consistency of the motor's reaction)
- the heads with later AKAI glass and crystal composition heads (as opposed to earlier alloy and ceramic cast heads) from another machine I'd taken apart. I now know this is

not a weak link in my system.

I've also replaced the following parts on my vintage Marantz 1060 (from 1971):

- all the coupling and filtering caps on the preamplifier

- the driver-stage transistors on the power amp
- the bridge rectifier with a new, 6A/200PRV
- the DPDT/TV-5 power switch with a newer Alps replacement. (I pried open the old power switch after removing it, and couldn't believe all the oxidation which had crept in.)

When the budget allows, I also plan to upgrade the output stage filtering caps (from Nichicon, VZ-Series general purpose) to really "elite" Cornell Dubilier, "MLP" Series Flatpack 50-year-life caps. (Digi-Key now offers these, and the ones I need are the 80V DC, 3300µF rating. . . at \$87.43 each!) Are they worth it? With all I've already done, I need only turn the volume up to the 8:45 position to hear it at full blast. This amp is supposed to deliver 60W per channel, and even with famously insufficient vintage Acoustic Research model

CONTRIBUTORS

Ed Simon ("Sound Strobe Review," p. 6 and "Speech Intelligibility, Part 2," p. 34) received his B.S.E.E. at Carnegie-Mellon University, and has installed over 500 sound systems at venues including Jacob's Field, Cleveland, Ohio.

Walt Jung ("Sources 101, Audio Current Regulator Tests for High Performance, Part 1: Basics of Operation," p. 10) is a noted analog designer and author of the classic *IC Op-Amp Cookbook*.

William Eckle ("Building a Center Channel for an Altec A7-800," p. 24) is featured on the website www.wmeckle.com.

Dennis Colin ("A Low-Noise Measurement Preamp" p. 26) has demonstrated the audibility of phase distortion at Boston Audio Society, and has designed the "Omni-Focus" speaker (bipolar coincidental with phase-linear first-order crossover), 1kW biamp and PWM supply at A/D/S.

Rudy Godmaire ("A Flexible Subwoofer Amplifier, Pt. 4," p. 40), a sales consultant for Bell Canada, has been interested in DIY audio since 1998.

Darcy Staggs ("DeNoising A Vanilla Preamp," p. 46) is a retired engineer. His work experience began with the Apollo project, then defense analysis, writing management software at IBM in Stockholm Sweden, and later more defense analysis back in the US. His later years as a consulting engineer involved product design, machine design, business development, and any other fascinating challenge that came in the front door.

Rickard Berglund ("Audio Aid: Build This Six-Transistor Low-Noise MC Preamp," p. 48) has worked in research in analytical chemistry, and in industrial electronics design. He is now engaged in construction and manufacturing of tube amplifiers.

2ax's (1968), I must be getting well over 50W per channel with no problem.

As for the recordings I'd like to mention:

1. *Arthur Fiedler: Superstar*. Boston Pops. Polydor 5008, 1970. Listen to the grand spectacle of their arrangement of the overture to Andrew Lloyd Webber's *Jesus Christ Superstar*. Also, an instrumental version of the 1969 country hit *Gentle on My Mind*, in which you should be able to clearly delineate the rhythm of the second horn section trombones from the glaring French horns in the first horn section. Even with just a two-channel recording, you should hear the echoic reverb of the percussion (due to the album having been made inside of Symphony Hall) on this track.

2. *Glory Of Love*. Herbie Mann, A&M 3001, 1967. A lot of jazz purists will dismiss this album as being "kitschy, lounge lizard pre-Fusion"; however, in one track, *Oh, How I Want to Love You* (composed by Mann himself), you should reproduce the kick and thump of Grady Tate's drums without any muddiness. Also, the extended guitar solo (by Eric Gale, whom I don't know whether or not he was famous at the time) should sound as though it's a Gibson flailing right in front of you. (The only way I can describe this song—which is an instrumental—is that it's a mix of a tango, the Pied Piper, and B.B. King style guitar thrown all together! People really took chances in the Sixties!)

3. *Revolver*, The Beatles, Capitol 2481, 1966. I think this is the best album the Beatles made (they were too imitative of other people before this point and became too gimmicky afterwards). On *Here, There, and Everywhere*, the sound-stage should clearly indicate the place of John and George's harmonizing (as well as finger-snapping) behind Paul's lead vocal. On *She Said She Said*, listen to how John and George each try to synchronize the lead and rhythm guitar parts on their respective Rickenbackers. On *Good Day Sunshine*, you should clearly pick up Ringo Starr mumbling something (I think it's "she does") after the first line of (I believe) the third verse, when Paul sings "She looks good."

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James Hoover
South River, N.J. aX



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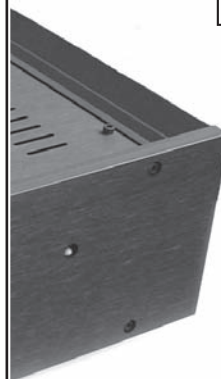
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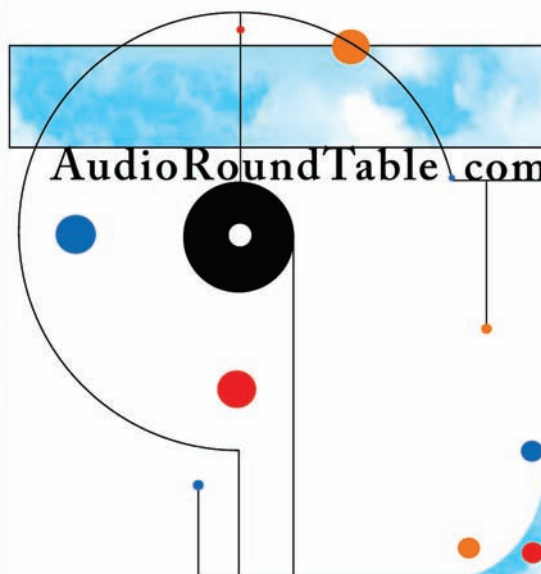


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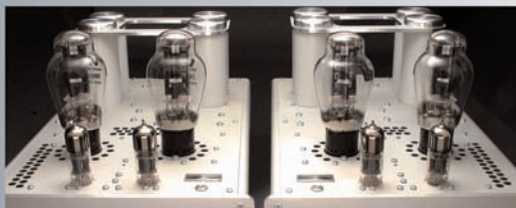
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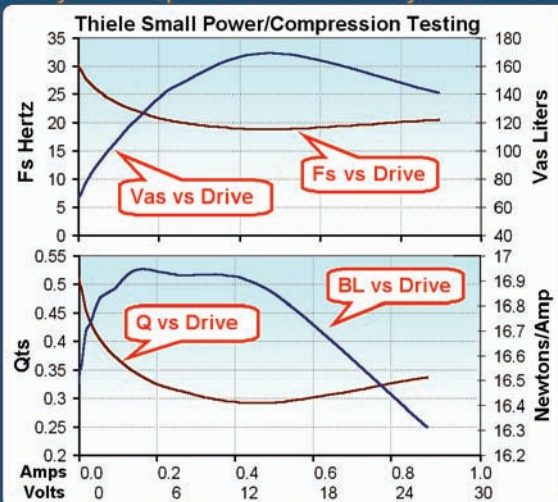
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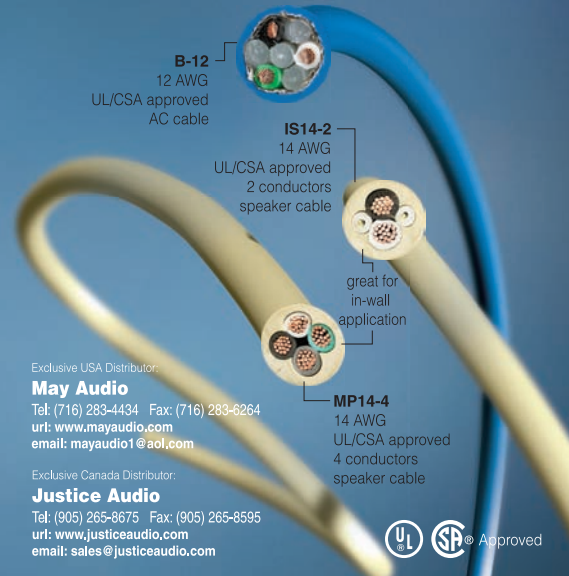
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