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Beginner's Project: BUILD THIS SE AMP

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A Beginner 6BQ5 SE Amp

Here's an amp project especially for first-time builders featuring the 6BQ5 tube.

PHOTO 1: Front view of the amp. (All tube model, unpainted chassis.)

In order for this beloved hobby of ours to thrive and survive well into the future, we must always strive to make it interesting to newcomers. These new hobbyists should be encouraged to explore the art of building their own equipment. This can be accomplished by providing them with simple, easy-to-build projects that will work as specified and will also give them the pleasure of knowing that they can complete it and proudly say that they built it themselves. These projects should be inexpensive to construct, should work from the very first time they are turned on, and also please the builder with their sound.

BABY BOTTLE KING

A friend of mine, Neal Haight, who has contributed many of his great ideas to this magazine, recently mentioned a lack of articles regarding what I consider to be the king of the "baby bottles," the 6BQ5/EL84 power tube. I have built some push-pull amps using this wonderful, hard-working little tube, but even though they were fairly simple in construction and low in parts count, I wanted to build one for first-time builders, and one that would sound really good.

The 6BQ5 sounds great in push-pull, but what would it sound like in single-ended triode mode? I had a few of these tubes in stock in my hobby room, so I dug out the parts I would need. I ordered a chassis from Antique Electronics Supply (tubesandmore.com) and found that the Hammond #H1444 was the perfect size at 12″ × 8″ × 2″. It held all of the components and has a nice open look, giving the tubes good spacing for keeping things cool (**Photo 1**). I had some Hammond output transformers handy, #1645, and the load of the primaries on these transformers gave me the necessary load for the 6BQ5s into my 8Ω speakers. I used the #BR-1 speakers from Parts Express, which, although not super high efficiency, sound really good with this amp.

I realize that PP transformers are not quite suited for single-ended circuits, because the SE units have an air gap in the core to keep it from saturating with DC. Using the push-pull Hammonds will cause a slight rolloff below about 50Hz, but that is close to the limits of the speakers I am using, so it does not cause any problems. Later in this article I will explain how to keep the DC from the primary windings for those of you who want a flatter frequency response. I didn't use it because, as usual, the engineers at Hammond are a bit conservative with their ratings, much as they were with the #125CSE transformers that I used in my "Mini Single-Ended Amp" project (*aX* April '04).

I could not detect much of a rolloff in the low end while listening to music, which, for me, is where it counts. Funny how the ears and the meters don't always agree, huh? You may use any output transformer that has the required load for the 6BQ5s, but the Hammond #125 series is a good and inexpensive choice to consider. Also, the wiring hookup for them is very easy.

I built two versions of the amp and I used tube rectification for the power supply in one and diodes in the other. You may use whichever you prefer. Just remember to use the correct type of input, a choke-capacitor input, so that, when used with the listed power transformer, your B+ won't be too high.

POWER SUPPLY

I used the 5V4 type of rectifier tube for the B+ in the amp because it was in my stock, will handle the load, and offers the time delay that I think is necessary, which keeps the high voltage from hitting the plates of the tubes before the cathodes warm up. Much discussion has taken place about the benefits of the timed delay of B+, so I won't go into it again here. Let me just say that I want to protect my NOS tubes as much as possible! If you use diodes for your power supply and you want the time delay, just use the octal (V1) socket for a delay relay #T-6C45-DR from AES. I used a timed relay that was in my stock. It has a certain "industrial" look to it, but it works very well.

If you already have separate transformers for the heaters (6.3V) and the high voltage (B+) that you can use for this project, you can wire in another switch to turn on the B+ after the heaters have warmed up for 30 to 45 seconds. There will be room under the chassis for most of the small transformers that are available from AES. Any one rated for 6.3V at 4A will do just fine. You can also wire in a switch and relay to turn on the B+ when using only one power transformer for the amp. In case you do use this switch and relay setup, then you can omit the octal socket altogether. Some of my other equipment has this switched type of "on-standby" and "operate" capability, which works well.

When using a timed relay or a switched relay to turn the B+ on, try using a 2W 100k resistor and a 100nF × 630V capacitor across the contacts to prevent the "thump" that you can hear in your speakers when the HV is sent to the tubes. The power transformer used here is from Hammond and provides all of the voltages needed in this amp without overloading. The part number is #T272HX. You will note that the measurements for all of the sockets and transformers' mountings are on the drawing for the layout of the components (**Figs. 3** and **4**).

Following the wiring diagram on the schematic (**Fig. 1**), you should have no problem with miswired connections. The 5V leads from the 272HX go to the filaments of the 5V4, pins #2 and #8, and the high voltage leads are connected to pins #6 and #4. Be sure to connect the B+ supply wire to the correct pin, #8.

The center tap of the transformer is connected to ground, which is where I placed the B+ protection fuse. Some surprises in life are fun, but a short in your B+ circuit is not one of them! This will prevent burning out a perfectly good power transformer and will save you the headache of having to buy another one. Remember, transformers are not usually returnable to the dealers when the windings are burned out, and, yes, they can always tell!

The type of filter used in the B+ circuit is of the choke input configuration. It places the choke in the wiring before the capacitor. I used two chokes in the power supply—one for the B+ to the outputs and another for the B+ to the input stage. There was room under the chassis and the chokes I had were correct for the loads without being pushed out of their range. You may use one larger choke if you wish, but you may need to adjust the voltages to the circuits.

Using these chokes makes it easier to obtain the required voltages for the amplifier circuits, and they fit under the chassis quite well. The resistances of the chokes and the other resistors keep matters in check. The 120V supply circuit is also fused for extra protection, a procedure that you should always do. The fuse ratings are in the parts list.

The heater circuit is straightforward in that it contains only a full-wave bridge rectifier and filter capacitor. Depending on your power mains line voltage at your home, you may need to adjust the capacitance to keep the heaters at around 6.3V DC (±10%).

Increase or decrease the capacitance accordingly. I used some standard electrolytic capacitors for the B+ filtering (47µF × 450V), which seemed to filter the B+ quite nicely. Be sure to observe polarity with these caps! I normally use bypass capacitors in the HV, and you may use them if you prefer to "tailor" the B+ to your liking. A 330nF × 630V

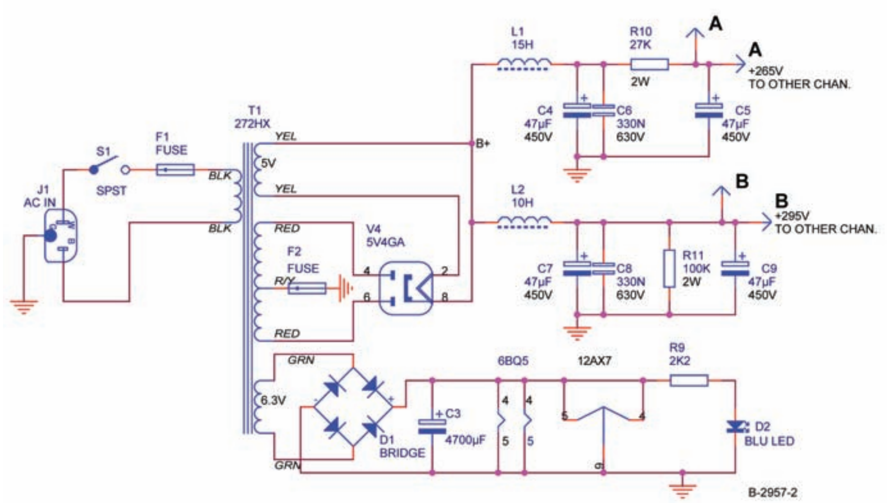

FIGURE 1: Vacuum tube power supply diagram.

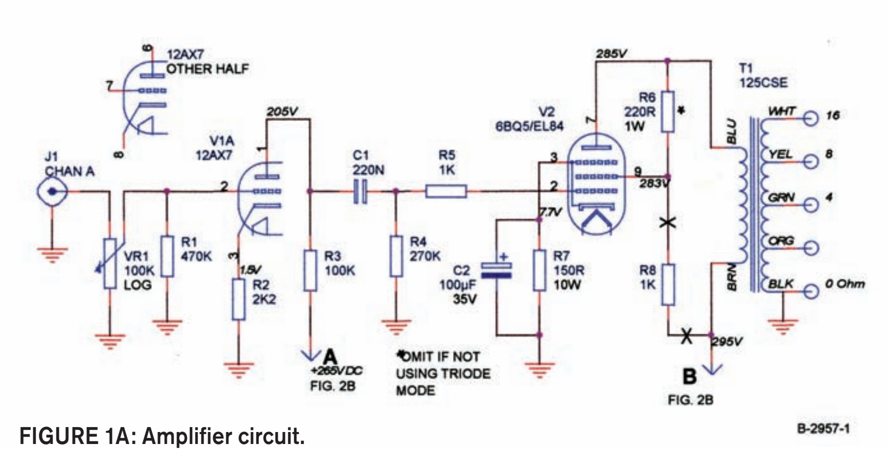

FIGURE 1A: Amplifier circuit.

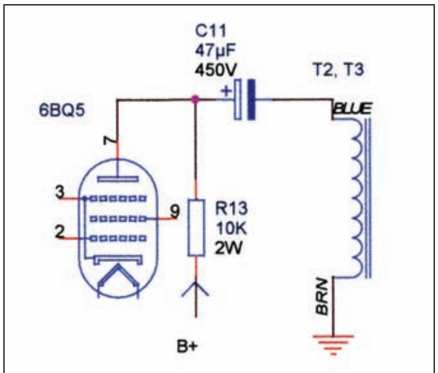

FIGURE 1B: DC blocking option if using push-pull transformers (Omit if using Hammond #125CSE.)

DC usually does the trick. The hum and noise in the amp is minimal, and you can only hear it when you place an ear close to the speakers.

If you decide to use diodes for your B+, then use the optional power supply diagram included with the schematic (**Fig. 1**). Use good-quality diodes. Both the standard 1N4005 and the high-quality Hexfred diodes are available from AES. When using diodes, you can add some extra filter capacitors to help smooth out your B+, which you can't always do when using rectifier tubes. I built two of these amps using both types of power supplies, and I can't really find much difference in the sound.

THE TUBES

I used a 6AX7 for the first stage of amplification. If you think that this is the same as a 12AX7, you are correct. The only difference is the heater voltage. I could have used the 12AX7 here, but I decided on some great NOS Tung Sol 6AX7s that I wasn't using.

I can't say enough about the Tung Sol tubes. I used their 6550s in my single-ended project (*aX* Sept. '01), and they remain my favorite brand. I realize that most of them are not available anymore, or are too costly, so I am listing other brands for this amp. There are many good 12AX7s available today to choose from. Just remember to wire the 12AX7 for the proper heater voltage.

You will note that each of the triode sections is dedicated to an individual channel. You may use either section for whichever channel you decide, or you may follow the pinout on the diagram (**Fig. 1A**). Take note: Following the diagram on any first-time project is *very* important! It will prevent mistakes that could possibly be costly and slow down the completion of your project!

The 6AX7 (12AX7) tube makes it easy to drive the amp to its full power. Using the circuit as shown with an input of around 0.1V drives the amp to a comfortable listening level when you use speakers that have an efficiency of around 90dB. And when you increase the input to around 0.5V, the amp is driven pretty hard and the distortion becomes evident. So, if you choose to use the amp in a direct-connect setup with your tuner or CD player, you will need some sort of volume control to prevent it from being overdriven.

In case you are not going to be using this amp with a preamp or other source with a volume control, then install the one shown on the diagram. If you want less gain on the input stage, you can experiment with a 12AU7 or a 12AT7. I'm using mine with a Dynaco CDV-1 tubed CD player with its own volume control.

The 6BQ5s—the power output tubes—are still easy to obtain, with many different brands to choose from. They deliver about 2W in triode mode, or, about 5W in pentode mode. These tubes are very rugged. Many guitar players still use practice amps that have them in either SE or PP configuration. Personally, I have never replaced a 6BQ5/EL84 due to failure; some have become a little soft sounding as they aged, but, hey, that's normal.

The tube uses cathode bias, which makes the circuit simple and allows you to try different tubes without needing to worry about any adjustments. I tried some JJs from AES that have a lot of mid-bass punch and a good bottom end. I also used some NOS Sylvanias which have a fantastic clarity about them. The midrange from the Sylvanias is great, and the tubes seem to cause the little amp to take on a personality of its own. I thought my "Mini Single-Ended Amp" sounded good, but, wow, the 6BQ5s are right at home running in SE triode mode!

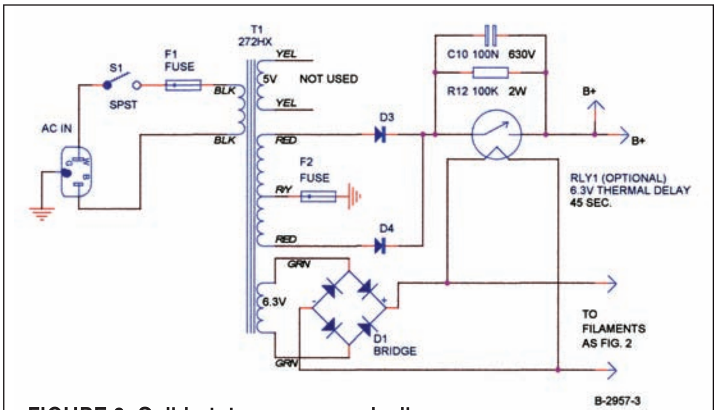

FIGURE 2: Solid-state power supply diagram.

Just how long can some tubes last? Well, just for the record I looked it up. It seems that there was a "Klystron" tube used in a radar installation once which accumulated about 240,000 hours on it. That's about 27 and a half years! I have pulled some tubes out of old console radios from the 1930s that still tested good.

After about a ten-hour break-in, the amplifier took on a certain "sparkle" to its sound, one that was very easy to get used to. The sound was effortless in its delivery and very easy on the ears, with no fatigue after a long listening session. I think I'm hooked.

CHASSIS LAYOUT AND CONSTRUCTION

The Hammond chassis box has plenty of room for all of the parts you will use to complete your amplifier. You will need a couple of hole punches for the tube sockets. The sizes are 0.75″ and 1.125″, or, in layman's terms, ¾″ and 1 1/8″. Or, you can have your local electrical shop punch out the pre-marked chassis holes with the Greenlee brand of punches that will do the job very neatly and quickly. Some hobbyists use a proper-size drill or metal nibbling tool such as the #ST806 that AES carries. Whatever works for you is OK.

When working with the chassis, be sure to remove all of the shavings and burrs that are left over from drilling and punching the metal. The spacing and hole sizes for the transformers and the placement distances are on the chassis drawing (**Fig. 3**). The mounting holes on your output transformers will vary, depending on which one you decide to use. Using this drawing will help take the guesswork out of where to put things on your chassis. Actually, you may arrange your amp however you wish, but I found the layout here works very well. You can see some good views of the amp layout in **Photos 1**, **2**, and **6.** I put the power switch on the right side because I'm right-handed.

I installed the blue LED on the left side. The glow of the tubes is usually enough to let you know that the amp is on, but I really like the little blue light on the front of the chassis. It's easy to tell the amp's status at first glance from across the room. This LED is very bright when used with the 6.3V DC supply in the circuit, so I used a resistor to reduce the brightness level. The resistor is 2.2k ½W and keeps the LED from being brighter than the glow of the tubes, which most audio hobbyists seem to enjoy.

I used #6 hardware for all of the mounting of the chassis top parts except for the transformers. I used #10 for them. The types of terminal strips in **Photo 3** are very handy for mounting all of your components and are available from AES or your local electronics store. You may use as many as you need to complete your chassis wiring hookups. You can mount them wherever you consider to be a good tie-in point for your wires and parts. The socket mounting hardware is always a good place for them because you will terminate almost everything into the tube socket connections. Try to route the wires carrying the AC power (120V) from your supply line away from your low-level signal wiring to reduce the possibility of any induced hum. Be sure to use rubber grommets on the holes that wiring will pass through on the top of the chassis and on the power wire hole in the back plate.

To make it easier to move the chassis around while doing the wiring, you might want to mount the transformers last after you are through with most of your hookups. This also makes it safer because the chassis is much lighter to handle. For the version with the diode type power supply, I painted the chassis (**Photo 6**) when I finished mounting everything. You can paint yours any color you like or leave it as is with the shiny aluminum surface, as I did for the version with the 5V4 tube rectifier power supply. Again, fellow hobbyist, have it your own way.

THE AMP CIRCUIT

The circuit used in this amp is very simple and easy to wire if you follow the diagram (**Fig. 1A**). You can include the volume control indicated on the diagram, and connected directly to the grid of the first tube, or omit it. The wiring for each channel is identical, and the easiest way to prevent mistakes is to wire a component, resistor, and so on, into one channel first, then repeat the process on the other side. Try to use heat-shrink tubing on the exposed component leads where possible. This will make your circuits much safer to test and will prevent any short circuits where you don't want them to happen.

After installing the part, look at the other channel to see whether they match and then compare both to the schematic diagram. Repeat this as many times as necessary until you are sure that all looks well. **Photo 3** includes a view of the inside of the chassis. The hookup of the output transformer is shown on the diagram according to the color codes for the Hammond #125CSE.

Depending on the type of transformer you use, be sure to follow the color codes indicated on the diagram that comes with your transformer. If you use the Hammond #125CSE, you will have fewer wires on the primary side. The #1645 has a center tap and two UL taps. I just isolated them, covering them with heat-shrink tubing, and secured them under the chassis. Depending on the resistance of your speakers, you will need to wire the secondary according to your diagram.

If you prefer to experiment with the hookups for the transformer, just remember to isolate and insulate the unused wires. You can use the extra circuit in **Fig. 1B** if you want to block the DC from the core of your amp's output transformer, but if you use a Hammond #125CSE you won't need it. Larry Lisle, whose work I have always admired, demonstrated this method in an article back in 1996 in *Popular Electronics*, where he showed how to construct an all-triode SE amp. I've never used the circuit, but if Larry says it works, that is good enough for me. If you don't already have some PP transformers on hand, then just order the 125CSEs and you can avoid the extra circuit.

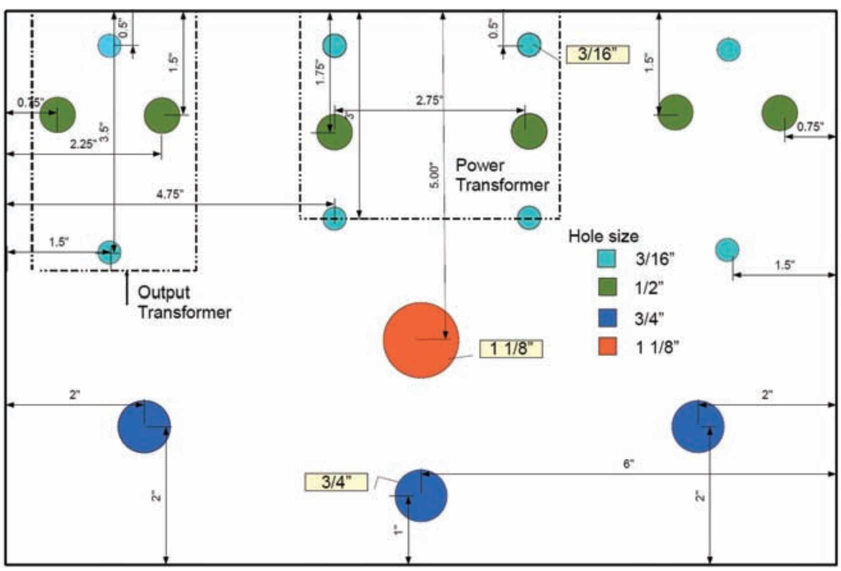

FIGURE 3: Chassis layout—top plate.

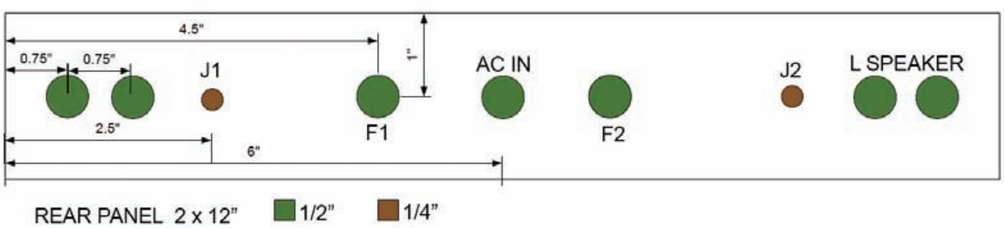

FIGURE 4: Rear panel.

One of the advantages of a single-ended amp is the low parts count, which means simplicity. Because it is single-ended, there is no need to use a phase inverter. I am not even using any feedback in the circuit. By using cathode bias for the output tube, you don't need to worry about a separate power supply for the bias. This is accomplished with the use of the resistor and capacitor on the cathode of the 6BQ5 to ground. I experimented with different values of capacitors for the bias and settled on the one on the diagram after extensive listening tests; again, tuning by the ear, not the meter.

By the way, I used "star" grounding and tied all of the grounds into a central point near the right channel input. With around 285V on the plate, I was looking for about 7.5V at pin #3 on the 6BQ5. I ended up with almost exactly that reading. I used 25W resistors for the bias because I had them handy. You can use a 10W for this tube because the heat output is well within that range. The 25W run really cool.

In **Photo 3** the amp is still under construction, and yet it should give you a basic idea regarding the placement of parts, but you can arrange yours in any way you wish. In order to use the 6BQ5 in triode, you must connect the plate, pin #7, to pin #9, as shown in **Fig. 1A** using a 220Ω 1W resistor. If you don't want to use triode mode but want more power, then connect pin #9 directly to the B+ going to the output transformer using a 1k0 resistor. This option is shown on the schematic diagram including the extra resistors. Personally, I like the triode mode better.

You may use any brand of resistors and capacitors to wire your amp, and, if it is your first project, you don't need to go for the "high end" parts. The parts listed have given me a really good sound and their costs are minimal. That is another one of the joys of this hobby: You can spend as much or as little as you like on your amp, you can modify it at any time in the future if you wish, and, if it ever needs it, even repair it yourself. The operating parameters for the amp are indicated on the diagram and can be easily measured with a standard volt-ohmmeter, either analog or digital. More about this in the testing section.

TESTING YOUR AMP

Before you start testing the circuits for proper voltage readings, you must remember that you have some high voltage under that chassis, both the mains power and the B+! The 6.3V and the 5V supplies will forgive most mistakes; the 120V AC and the B+ will forgive none! If you are not used to measuring these types of voltages, you should do everything possible to ensure that you will have a safe and satisfying experience learning how.

If you don't have a test meter, these are available from AES or from your local electronics store. Read the instructions

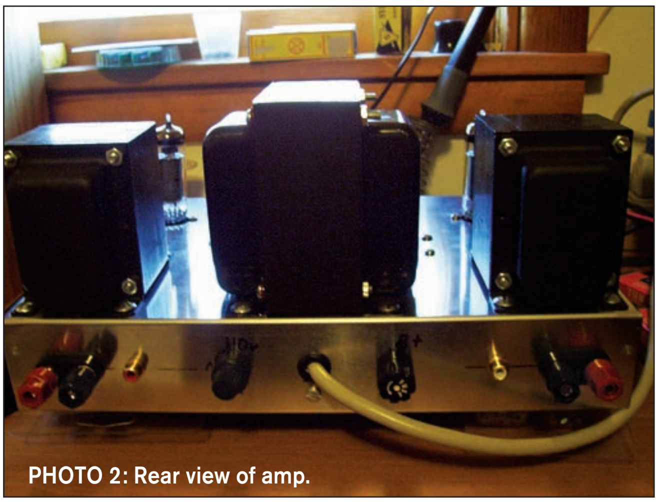

PHOTO 2: Rear view of amp.

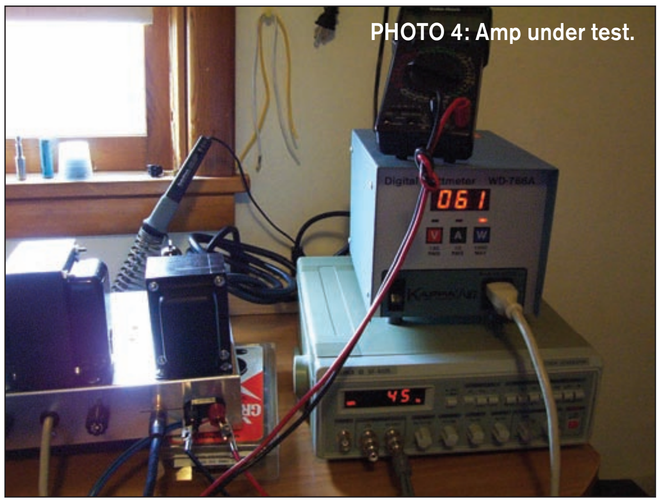

PHOTO 4: Amp under test.

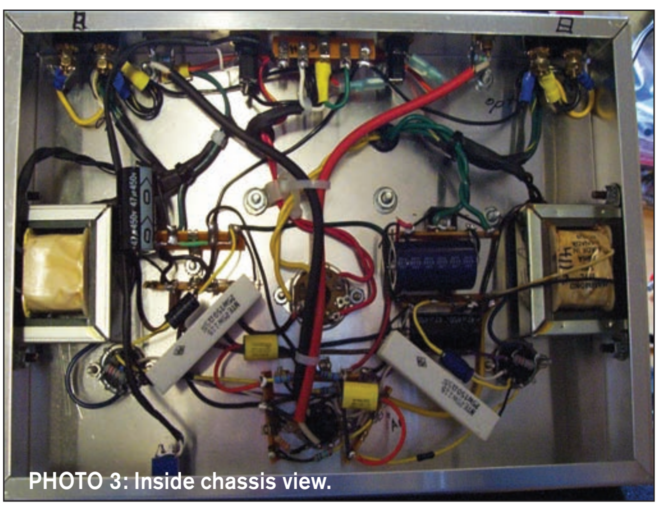

PHOTO 3: Inside chassis view.

PHOTO 5: A safer test probe.

that come with your meter and practice measuring with it on some low power circuits such as batteries, resistors, and so on, until you feel comfortable with its operation. Remember, your eyes can only see the components of your amp, the meter will tell what is actually going on inside of the circuits.

In **Photo 5** you will see a standard test lead probe that has been made much safer. I slipped some pieces of heat-shrink tubing over the shaft and shrank them with a heat source, leaving only a small portion of the tip bare at the end. This tubing is rated for 600V and will keep you from shorting the probe to ground or other connections while touching the test points on the circuit. This will keep you and your amp much happier.

Because all of the readings are made with reference to ground, use a clip to secure the black negative (-) probe to the chassis ground so that you don't need to hold it with your other hand. This way you won't lose that sparkle you have in your smile when nearly 300V of power passes through your body! Touch each test point with your red positive (+) test lead while holding it with *one* hand only. Many experienced hobbyists will put the other hand in their pocket while testing—a wise move.

Note the reading and write it down if you wish. Your readings will be within about 5% of the readings on the diagram, depending on your power source. Leave the fuse out of the B+ fuse holder when reading the low voltage points. Again, safer!

The 6.3 and 5V sources may appear a little high without the load of the tubes on them. This is normal. After you are sure that you have the correct voltages for the filament and heater supplies, unplug the amp and install the tubes. Push them into their sockets carefully, without forcing them.

Put the B+ fuse back into its holder and connect your speakers to the proper terminals, observing the polarity. Plug the amp back in and turn it on. Watch for the soft glow of the heaters on the 6AX7 (12AX7) and the 6BQ5s and the glow of the filament on the 5V4 (unless you used the diodes for your power supply).

You may hear some hum coming from your speakers as the amp warms up. When you touch the input jack center lead, that hum will become louder. This is normal. If you don't want to hear the pops and hum produced while testing, simply short the center of the input jack to ground. This type of short is acceptable. After the amp warms up, take your readings of the circuits again. Remember, now the B+ is flowing!

Your heater, filament, and B+ readings should all be close to normal. Sometimes in my area the utility power voltage will go as high as 125V and the B+ in my amp will hit around 300V. This doesn't really cause any problems because the tubes are rated for 300V. They will become slightly warmer and I notice a little increase in the volume, but that is not serious. If you notice that a bad solder joint is making a problem, turn the amp off, unplug it, and wait for the capacitors to discharge.

Note: The B+ capacitors are discharged by the load of the 100k resistor R11. The filter capacitor for the heaters is discharged by the constant load of the heaters. Resolder the joint. If the joint is on a tube socket, remove the tube before applying heat to the socket terminal! After you are certain that everything is OK with your amp, you are ready to hook it up to your preamp and/or a music source. I used some stick-on rubber feet to give the amp some air circula-

PHOTO 6: Front view, painted chassis. Solid-state power supply version.

tion under the chassis. This is always a good idea.

HOW DOES IT SOUND?

At first I did some frequency response tests from the amp (**Photo 4**), but that is never close to the dynamics of real music. I played a track from Toni Braxton, "Spanish Guitar," and noticed how the sounds of the strings—actually each individual string—of the instrument seem to come forward from the speakers until I felt as though I could reach out and touch them. I played a variety of music from classical to soft rock, to country, and some oldies but goodies, and I started to hear things that I hadn't noticed on other amps, especially solid-state.

I built a 25W per channel SS amp, and I swear it is as though it places a veil over the midrange and upper midrange. I have modified it to the hilt, but it just won't sound natural. This little triode amp makes it sound as though the music coming from its silicon parts is smothered under a mattress. Voices from this triode amp take on a new life, even at low volume settings. The voices of Hayley Westenra and Enya are almost unreal. The soft violins on "Clair De Lune" and Martha Babcock's cello on "The Swan" from Boston Pops with John Williams almost brought a tear to my eye. Piano music seems very natural.

So, I wonder, how can this simple, inexpensive amp stand on its own with some of my other equipment? There are very few parts in the signal path, and the triode sound is virtually unmatched in audio amps. I listened to a 300B amp once, and, let me tell you, that will spoil your ears real fast. I realize that this amp is not a "true triode" amp, because I'm using a pentode that is strapped into triode configuration, but I think that the little 6BQ5/EL84 is happy running in triode mode.

I also didn't use any feedback in the circuit, because I believe that feedback would actually take away from the SE triode experience. Feel free to experiment with it if you desire. Mine sounds fantastic just the way it is. Now I have a whole new tube amp to listen to in my hobby room.

By the way, the amp draws 61W at idle and around 64W at full power, so it is very easy on the utility bill. Remember to give your amp a few hours to settle in, and you will start to notice the warm yet lively sound it can produce.

If you are a first-time builder, remember to take your time as you go through this project. Don't rush. To hurry can only produce mistakes, while taking your time will cause you to absorb the reason and purpose of your endeavor. This way you will learn and understand more about vacuum tube circuits. If you desire to learn even more about the "why" and "how" of tube equipment, obtain some of the books available from Old Colony Sound Lab found on the audioXpress.com website.

The tubes and other parts listed for this amp are low in cost yet will give you a finished product that will be well worth your effort. I'm sure that there are others who will want to modify and improve on this amp's design, but I am going to leave mine as is, sounding great!

I hope that your version of this amp pleases you with its sound. I also hope you find the same joy in building your own equipment as many *aX* readers. If you do decide to try this as a first-time project, let me be the first to welcome you to the Do-It-Yourself Audio Club. *aX*

PARTS LIST

Reference	Part type, number, and quantity	Source
Chassis box	Hammond, #P-H1444-22, 12″ × 8″ × 2″, 1 each	AES
Tube sockets	Octal, #P-ST8-808, 1 each, 9 pin, #P-ST9-211, 3 each	AES
Fuse holders	Chassis hole mount, #S-H201, 2 each	AES
F1	2A fuse, slow blow, #270-1023, 1 each	Radio Shack
F2	½A fuse, slow blow, #270-1061, 1 each	Radio Shack
S1	toggle switch, DPDT, 125V, #275-663, 1 each	RS
J1	input jack, RCA type, #274-346, 2 each	RS
SBP	speaker binding posts, #S-H263, 2 each	AES
T1	power transformer, Hammond #P-T272HX, 1 each	AES
T2, T3	output transformer, Hammond #P-T125CSE (or equivalent), 2 each	AES
L1	choke, 15H, 75mA, Hammond #PT-158L, 1 each	AES
L2	choke, 10H, 100mA, Hammond #PT-158M, 1 each	AES
V1	rectifier tube, #T-5V4GA, 1 each (omit if using diodes)	AES
V2	twin triode tube, #T-12AX7-S-JJ, 1 each	AES
V3, V4	beam power tubes, #T-EL84-JJ, 2 each	AES
TDR	timed delay relay, #T-6C45-DR. 1 each (if using diodes)	AES
C4, 5, 7, 9, 11	47µF × 450V capacitor, #C-ET47-450, 4 each (6 if DC blocking)	AES
C1	coupling capacitor, 220nF × 630V, #CFSD22-630, 2 each	AES
C2	100µF × 35V capacitor, #272-1016, 2 each	RS
C10	capacitor, 100nF × 630V, #C-TD1-630, 1 each (for relay contacts)	AES
C3	4700µF × 35V capacitor, #272-1022, 1 each	RS
C6, C8	capacitor, 330nF × 630V, #CFSD33-630, 2 each (optional bypass)	AES
VR1	dual volume control, #R-VA-8mm-2A, 100 k, 1 each (if you prefer separate channel balancing use 2 single controls)	AES
R1	470k ½W metal film resistor, 2 each	AES
R4	270k ½W metal film resistor, 2 each	AES
R3	100k ½W metal film resistor, 2 each	AES
R2, R9	2k2 ½W metal film resistor, 2 each (3 if using LED)	AES
R5, R8	1k0 ½W metal film resistor, 2 each (4 if not using triode)	AES
R11, R12	100k 2W metal oxide, 2 each (extra is for optional delay)	AES
R6	220Ω 1W metal oxide resistor, 2 each for triode mode	AES
R10	27k 2W metal oxide resistor, 1 each	AES
R7	150Ω 10W power resistor, 2 each	AES
R13	10k 2W metal oxide resistor, 2 each (if using DC blocking)	AES

Note: The above listed resistors are found on the "tubesandmore.com" website for Antique Electronic Supply. Just match the value and quantities of your resistors to their numbered codes.

D3, D4	3A, 400PIV diodes, #276-1144, 2 each (diode power supply)	RS
(D1) FWB1	6A, 200V full wave bridge for heaters, #276-1181, 1 each	RS
(D2)LED	blue, ultra bright, 1 each, various type and styles	RS
power cord	standard 3 wire, #16 gauge, #S-W125, or equivalent	AES
knobs	various volume control knobs	RS and AES
terminal strips	4 and 5 lug type, quantity is your choice	RS and AES
hardware	screws, washer, and nuts, #6 and #10 sizes, sufficient quantity; rubber grommets for wire protection, ½″ hole size, 7 each, rubber stick-on feet, 4 each	most local hardware stores
Miscellaneous	hookup wire, various sizes and colors, sufficient quantity for all circuits, heat shrink tubing for bare component leads and test probe modification, shielded audio cable for input circuits, solder, and so on	local RS stores

 By J.R. Laughlin

An Automated Level Control

Here's an inexpensive, easy-to-build sound level control.

PHOTO 1: The finished audio level control.

This circuit contains an audio amplifier capable of driving a speaker with circuitry that automatically maintains the same relative maximum output regardless of extreme levels that are occurring at the input. It does not "compress" the peaks, but simply "turns down the volume" as you would do with your volume control (**Photo 1**).

This maintains a steady output volume for very widely varying input levels. Extremely low distortion results from the use of a self-adjusting, purely resistive voltage divider circuit to perform this action, an LDR (light dependent resistor), instead of a semiconductor device. The distortion produced by the LDR was unmeasurable with my H.P. 5L4N spectrum analyzer. I thought that the particular voltage divider used here was an interesting approach to perform this. I originally designed the circuit to be used with an old audio oscillator whose output level varied considerably with changes in frequency. It cured that problem beautifully.

I also have applied it to the following:
- regulating TV sound level (loud advertisements!)
- my front door intercom
- CB radio and amateur radio sound regulation
- a public address system
- music systems

The circuit (**Fig. 1**) is very inexpensive to build and the operational current level quite low, making battery operation feasible. Several of my friends have built their own and they have all really enjoyed putting it to use. I am certain all you readers will also.

CIRCUIT OPERATION

U1 receives the audio. It has adjustable gain with R14, which has pads to accommodate a fixed resistor or a variable one. You can use R19 to adjust input level. C10 is useful with R18 for elimination of RF (if present) and also to adjust high-frequency rolloff. You can use R26 for adjustment of input if needed.

The op amps are operated with positive supply voltage only; they must have the inputs biased to approximately half of the supply voltage. This is done using R29 and R31 to supply the half voltage, which biases pin 3 of U1 through R25. The DC output of U1 is also equal to this half voltage; because of C5, the DC gain of U1 is only 1. Note that this half DC voltage is transferred to the input of U2 through R12 and 13, providing bias for it also (U2 DC gain is 1 also, C2).

Input bias current of U1 and U2 is so very small that the resistors do not have any important effect on the DC bias level. Audio out is present at pin 6, U2, or from the output of U3, which is used as the speaker driver. You can connect the input to U3 to R5, R8, or to C1, and adjust the U3 (LM386) gain over a range of 20 to 200 using R27. It will deliver approximately 0.4W to 8Ω with an 8V supply.

You can obtain P-P output voltage close to the DC supply level from the LM386 output for resistive loads from approximately 100Ω up. Note that the power supply is very simple and conventional. You can use D3 with DC power to prevent any problems if you accidentally apply a negative supply voltage to the input terminals. Of course, it is needed for AC power input.

HOW IT WORKS

The use of U4 is optional; there is really no need for it if you use an applied voltage that is compatible with the ICs. Here is a simple explanation of how the leveling is accomplished:

1. Audio from the INPUT terminal goes through U1, with adjustable gain.
2. This audio is amplified by U2, gain set by R4.
3. If the audio amplitude is large enough from U2, it is clamped positive by C3 and D2 provided that SW1 is closed.
4. This positive clamping action can approximately double the positive amplitude value of the audio applied to D1.
5. D1 rectifies this audio, which is filtered and smoothed by C6. And, if this DC value is large enough, it will cause Q1 to begin conduction.

6. Current through Q1 causes the LED to illuminate.
7. The light from the LED falls upon the Cds cell, which causes its resistance value to decrease greatly.
8. Note that the Cds resistance and R12 form an AC voltage divider (only when the Cds cell is illuminated; otherwise, R-Cds is too large).
9. This voltage division reduces the AC voltage amplitude applied to the input of U2, reducing the voltage applied to D1 rectifier and Q1 gate.
10. The result is that when the U2 output reaches a level sufficient to turn on Q1, then further increases to the U2 input from U1 cause the LED to brighten reducing the ohmic value of the Cds cell providing more voltage division to the input of U2.
11. One way to put it simply, we have a "disagreement" going on here. U2 is supplying more voltage to a circuit that is acting to reduce U2 input.
12. So, the maximun average output of U2 is automatically adjusted to a relatively fixed level when an input level begins to turn on Q1.
13. The input amplitude at which leveling begins basically is determined by the total gain from INPUT to output of U2.
14. The purely resistive and linear characteristic of the Cds cell causes no distortion of the audio.

ADJUSTMENTS

You can adjust the low- and high-frequency rolloff characteristics of the audio with time constants of R18-C10, C5-R15/14, C8-R13, C2-R3/4, C1-C4-R5, R27-C16, C14 and speaker. R9 and C6 affect the averaging of the rectified audio and how fast the circuit responds to an increase in amplitude. R10 and C6 determine how long a rectified and smoothed DC amplitude value remain after a decrease in audio amplitude.

There are quite a few potentiometers and components put into this circuit for experimental purposes, if you desire to do so. You can use R23 to alter the slope of the output versus input level. **Table 1** graphs the equalization for a U1 gain of 1 and U2 gain of 5 with a value of R23 = 0, 330, and 1K.

Table 3 gives a listing for **Table 1**. **Table 2** shows various AC voltage levels for two different input levels at T1 (level 2, equalization has started). I never use R21, but it is available if desired. R20 was originally used with a Darlington transistor and is not needed with the FET. Any of the variable pots can be panel mounted. You can use R22 to reduce input to U3 if needed.

There are nine pads located on the circuit board where you can place vertical posts to serve as easily available test points if desired (T1 through T9). These pads are numbered on the top of the circuit board. I used some very small brass nails for these test points. Refer to the *aX* website, www.audioXpress.com, to see the printed circuit boards and component placement.

I was interested to see how test results showed what a small amount of LED current would cause such a large amount of resistance change in the LDR (**Table 1**). Measuring the LED current (T4 and T5), I noticed on my unit that the audio voltages at C8 and T2 were the same

Table 1: Cds resistance vs. LED current (an average of several units).

20µA	600K
30	320
40	250
50	180
60	140
70	120
80	100
90	80
100	70
200	30
400	13
500	10
100	4.5
200	2.2
4000	1.2

Table 2: Audio voltage measurements.

2 input levels at T1, RMS		
	level 1	**level 2**
	.100	.288
Resulting levels at T3, RMS		
	.450	.597
T6	.734	1.21
Clamped peaks at T7		
	+.936	1.42
	-.232	-.262
LED current (R24)		
	.16mA	.28mA

Table 3

R23-	0	330	1kΩ
Ein(T1)mV		Ec(T3)mV	
100	450	450	450
200	584	597	619
300	600	623	665
700	625	680	775
1200	640	726	866
1400	645	741	896

level when the LED current was zero. When the LED current reaches approximately 0.03mA on my unit, the audio voltage at T2 is 10-20% lower than that at C8, showing that this small amount of current through the LED has started the process of reducing the LDR resistance, which is beginning to control the output from U2. These numbers will vary, depending upon the particular LED and LDR used.

The input from your application will determine what gain is appropriate for U1 and/or U2 or what attenuation at the INPUT terminal will be needed to begin the leveling at your desired amplitude. Of course, you can set the gains for small signal operation and adjust the input with R19 for larger inputs. Pads are provided for all potentiometers so that you can easily use a fixed resistor or two in place of any one.

R18 and R26 can serve in place of the potentiometer R19. Vertical mounting for R5 replacement and for R22 is required. Actually, for practically any application I can think of, unless the input is extremely small, you can permanently set the gain of U1 to 1 and use R19 and U2 gain to produce the desired amount of signal. The approach here is very flexible. This circuit works nicely as a speaker driver or as an input device to any other piece of equipment.

CONSTRUCTION

This is easy and typical. In the event, for some reason, that you might prefer a single-sided board, you can use four wire jumpers to provide the traces on the top-side of the board; nothing else on top is vital or necessary to the operation, only convenient. To me it is vital to provide easy removal and installation of parts you may want to experiment with. I really like the female "machine contact" (**Fig. 2**, sockets—Jameco # 102201, for instance). You can carefully clip the plastic from the edge of each one to free it. A 1/16″ drill bit will allow easy insertion into the circuit board.

I also prefer to use male and female crimp terminals for easily disconnecting wires. **Figure 3** shows how I prefer to do this. A bit of silicone grease really makes insertion of connectors easier. You can see in **Photo 2** several wires provided with these connectors.

In the parts list (**Table 4**) I suggest several hardware components, but your choice of these is wide open. I used the small die cast case; if you do likewise, you'll need to be very careful about hardware mounting because of the tight fit. The circuit board has a mounting pad on each corner; these

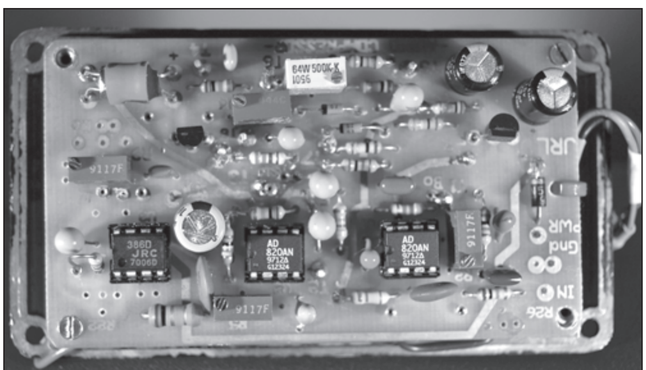

PHOTO 2: PC board.

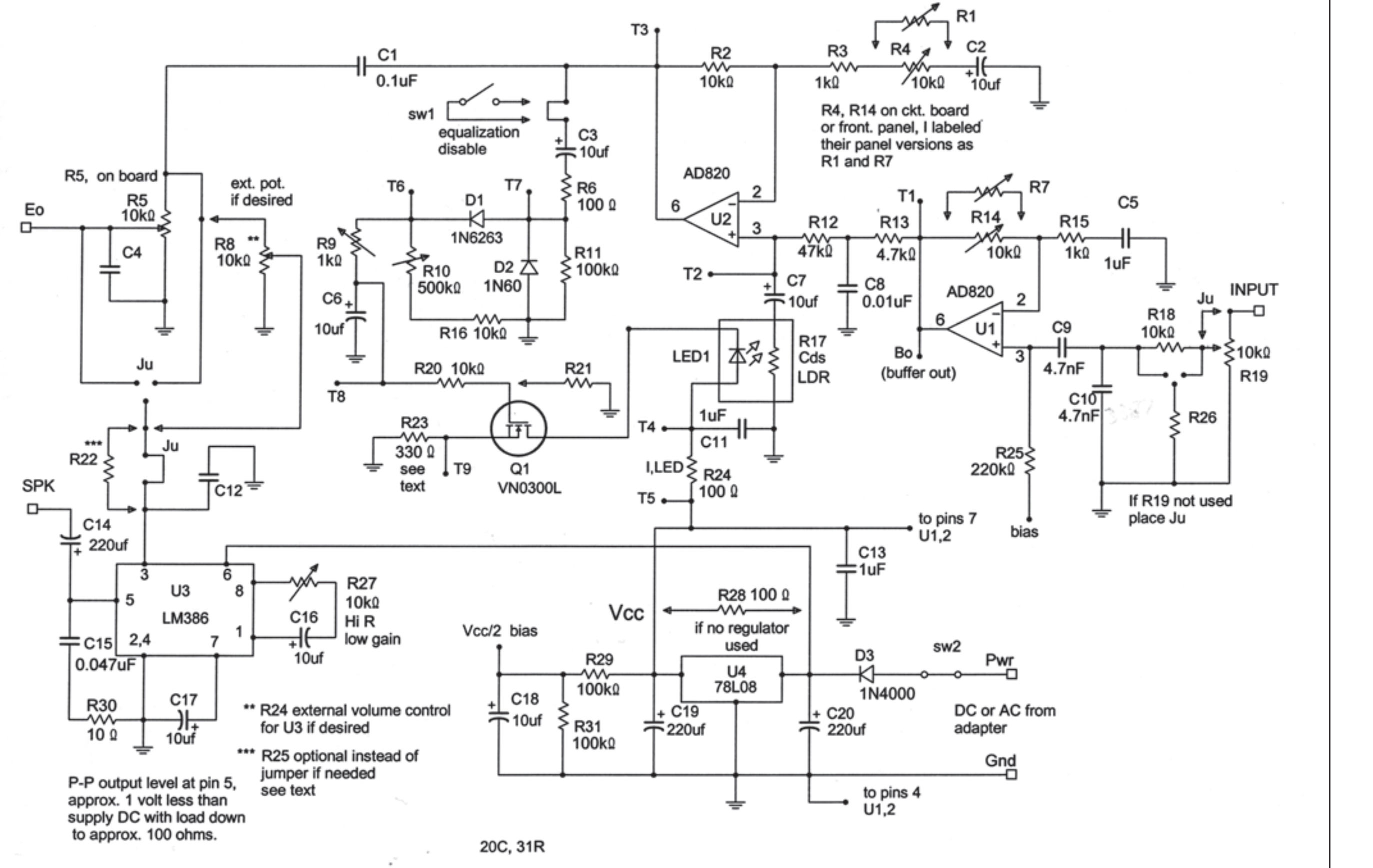

FIGURE 1: Audio level control circuit.

were basically designed for #2 bolts and standoffs to fit your cabinet. When you obtain the CDS cell/cells, test their resistance in *total* darkness; it should be over a megohm.

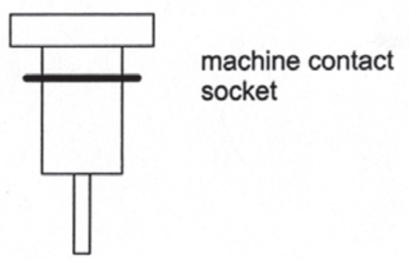

FIGURE 2: Sample socket.

When putting together the LED/CDS unit (**Fig. 4**), be sure to do a good polishing job on the flattened LED front. Also, be certain that the clear epoxy glue is completely distributed all over the space between the two surfaces of the LED and CDS. It is important that no light be able to enter the casing and affect the CDS cell. I used a drop of nonconducting silver paint placed into each end of the unit after the shrink tubing is in place. **Table 2** shows an average of measurements I made on several CDS units.

TESTING

My measured power supply current is approximately 6mA using an 8V regulator. Using a 100Ω resistor in place of the regulator, my current level is approximately 4mA with input supply voltage from 5 to 15V. This can depend upon the op amps used. My LM386 uses about 3mA of this total, so obviously if you do not need the LM386 or voltage regulation, do not use either and the static current drain will only be around 1mA for battery operation.

I suggest that you measure the output pins of U1 and U2. This voltage should be close to half of the supply voltage, which is determined basically by R29 and R31. **Table 2** has voltage listings for two different input levels of 1kHz sine wave

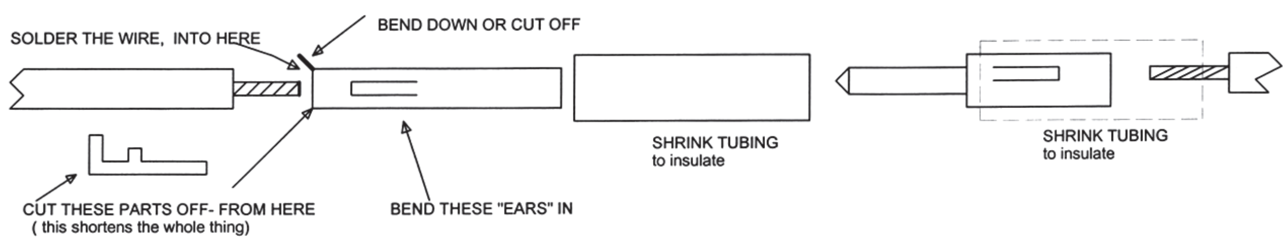

FIGURE 3: Preparing crimp terminals for connecting two wires.

starting at T1. Level 1 shown is below the level where Q1 begins to conduct, so all circuit voltages are the same as though SW1 is open. Level 2 is high enough so that the equalization process is in effect. Run a test as shown in **Fig. 2** and compare the results.

Test with varying sine wave input levels (1kHz) and observe when the equalization level begins. Test with music and voice over many amplitude levels of input while turning SW1 on and off noting the difference in output. I used a value of 330Ω for R23. This causes U1 to turn on a bit more smoothly and seemed to make the voice and music sound better. See what you think. *aX*

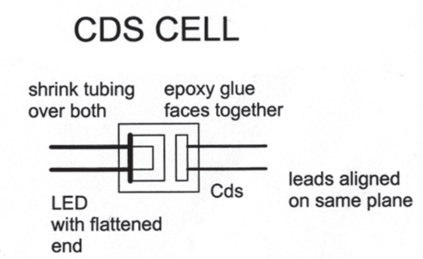

FIGURE 4: LED/CDS connection.

Table 4: Parts List

Capacitors	
C1	0.1µF
C2, 3, 6, 7, 16-18	10µF
C4, 12	see text
C5, 13	1µF
C8	0.01µF
C9, 10	.0047µF
C11	not needed
C14, 19, 20	220µF
C15	.047µF
Semiconductors	
U1, 2	AD820 or similar
U3	LM386
U4	78L08
D1, 2	any small Schottky diode
D3	1N4000 series or most any silicone diode
LED1	yellow hi intensity
Q1	VN0300L FET or similar
Hardware	
Switch 1,2	small SPST
Cabinet	Die cast LM Heeger 3421 or Jameco 11965, wide choice here
Audio connectors	RCA jacks, panel mount, available many places
Power connector	many options, Jameco 281851, 297529 or similar
Panel knobs and labels	custom
Power module	similar to Radio Shack 273 1767, wide choice on this
Machine contact sockets	Jameco 78642
Crimp terminals	Jameco 224581 male, 224573 female
Resistors	
R1, 2, 4, 5, 7, 8, 14, 16, 18, 20, 27	10K
R3, 9, 15	1K
R6, 24, 28	100
R10	500K
R11, 29, 31	100K
R12	47K
R13	4.7K
R17	Cadmium sulfide LDR–Radio Shack 276-1657 or equivalent
R21, 22, 26	see text
R23	330 (*Fig. 2*)
R25	220K
R30	10

A Greek Triad

With its unusual shape, this little tube amp—dubbed the Wood Star—shines brightly.

PHOTO 1: The completed Wood Star.

The idea to build a little SE tube amp was my friend's, Paris K., who worked for many years as a contributor to *Sound and Image* magazine here in Athens. For years, he begged me to make him the smallest SE using only a tube, if possible. Then I built him "The Mosquito," a 2W amp with one ECL82 (6F3P Russian or 6BM8 American equivalent) on each channel.

Paris did not want any special look, so I made the Mosquito on two thin plates of aluminum—1.5mm mounted one up and the second down, linked by four long screws. I took all the parts from old TV sets, with special attention given to the two little output transformers, which I treated seriously. Tim Giatras made them to meet our specifications.

Astonishingly, the little unit worked fine, with different sensitive speakers,

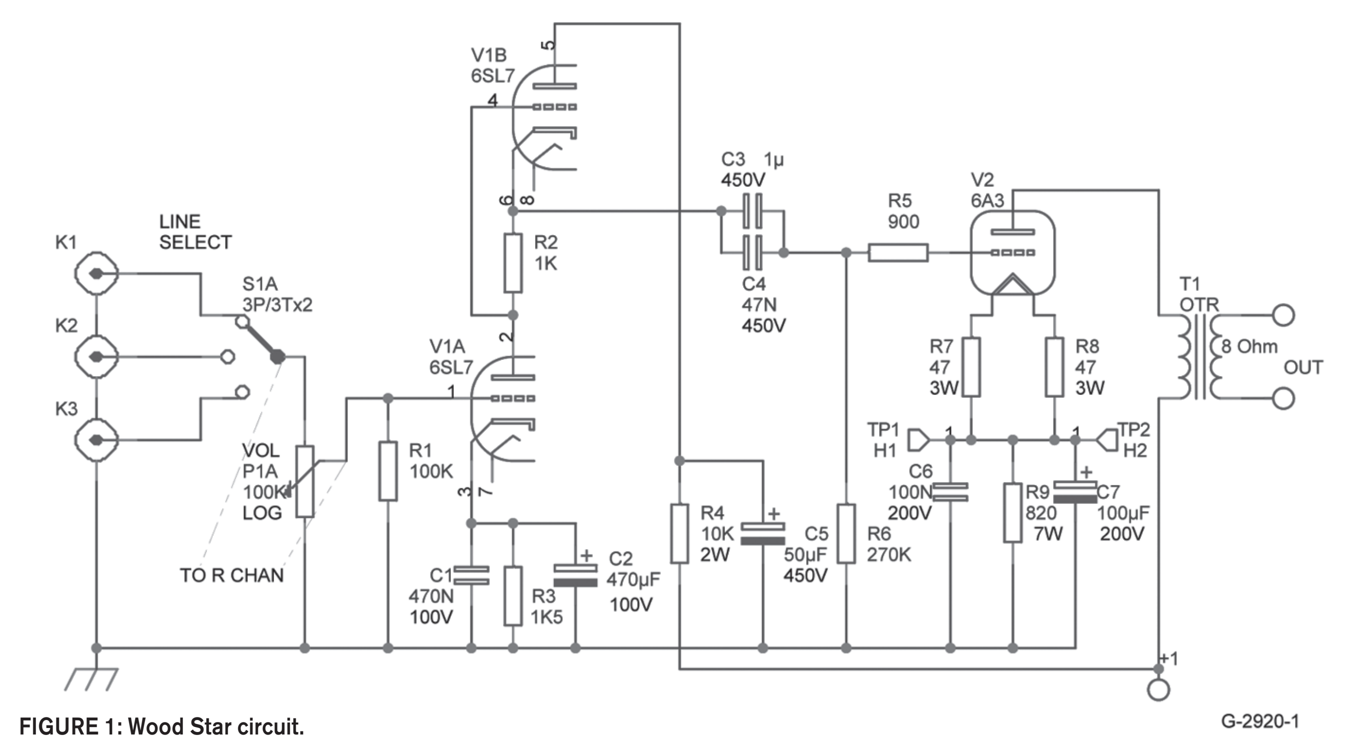

FIGURE 1: Wood Star circuit.

and is still working. Then, all the team assembled in my little lab to brainstorm. After consuming some bottles of Uzo, we finally hit upon an idea for better construction.

THE PLAN

We dubbed our next amp—due to its shape—"Wood Star." It includes the following features: no feedback, three channel line type inputs, volume control, and all the transformers hidden inside the chassis, with a big toroidal power transformer, in the upper leg, and the two output transformers inside the other two legs.

I chose a pair of 6A3s I had in stock (RCA) and two 6SL7 (6H9C Russian) as preamplifiers, because the 6SN7 in SRPP configuration didn't give enough "attack" to the power stage. Referencing many *GA* and *aX* articles, I finally came up with the circuit schematic (**Fig. 1**).

As you can see, there were no complications and no crazy ideas in this conventional design. There also was no feedback and no hum because of the special mounting of all the transformers

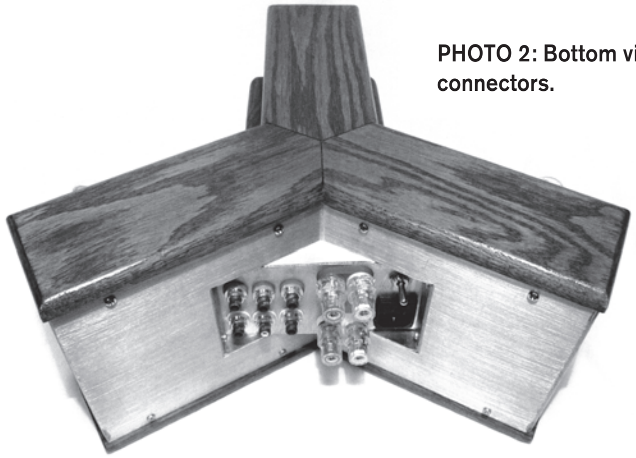

PHOTO 2: Bottom view showing connectors.

PHOTO 3: This versatile amp can be oriented horizontally or vertically.

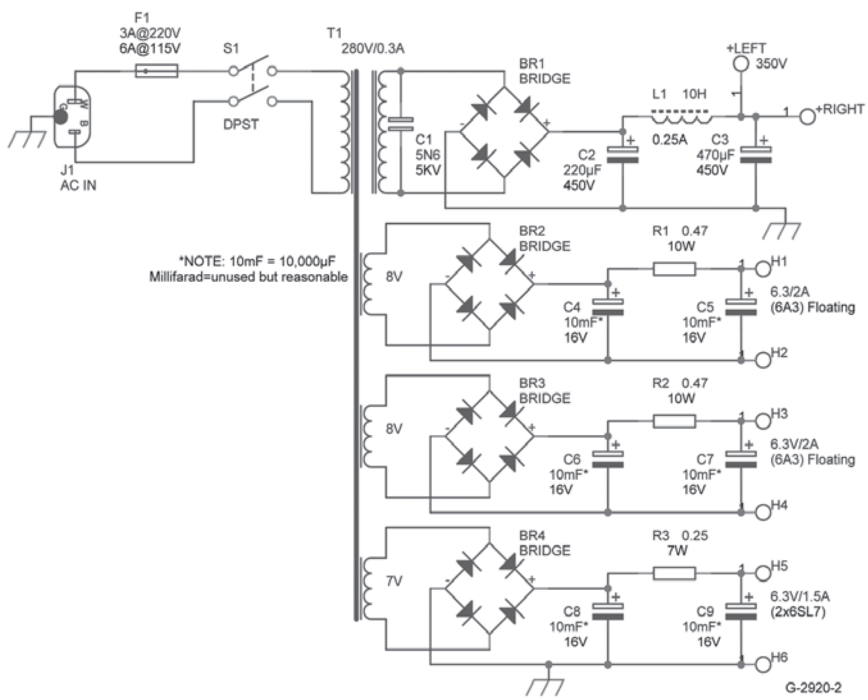

FIGURE 2: Two tubes per channel.

and the very good quality of all the parts used. For the power supply, I used superfast diodes in all the four bridges, and for the high tension I did not use a tube rectifier because I promised to use only two tubes on each channel (**Fig. 2**).

Due to the big toroidal power transformer (Tim), and to the special electrolytic capacitors (Nippon Chemicon), I didn't find it necessary to stabilize another power source, but I rectified all the heater supplies. Hristos D. provided me with massive silver connection wire (for the signal paths only); Aris made the technical design; George P. dealt with many problems in the beginning; and I began to solder the RCs on the isolated military supports. As I said, we used only expensive and brand name parts, such as Alps pot, WBT connectors, Multicap caps, and Caddock, Siemens resistors. All these parts were wired point-to-point.

RESULTS

The results were pretty good for a minimal SE: 3.5W RMS, an 8Ω charge, and 22Hz to 65kHz. With ±1dB band of frequency, the input sensitivity was 1.2V RMS for all three line inputs. The general aspect was made with six oak pieces, cut as in the photos. The amp can work in two positions—vertically or horizontally. The knobs are old-fashioned (American radio transmission), and the in/out connectors are located down.

During construction, I experimented with different input tubes, such as 12AX7 (6H2P), and E82CC (12AU7), and the Star worked well enough with minimal parts changes, but the general "look" was not the same—the old-fashioned 6A3id wasn't matched with the modern miniature noval tubes!

In summer 2006, I carried the Star to *Sound/Image* magazine, and after the tests the staff fell in love with it. The mechanical and electronic construction was not easy, due to the strange space inside the three legs, but the sweet sound of 6A3 was ample reward. The WE-A3 has the best bass sound, but even with Chinese or Svetlana tubes the Star was working OK—with, of course, easy speakers, up to 90dB sensibility. Those willing to tackle this project can get detailed technical data for the chassis and all the mechanical data from me at arcosound@hotmail.com. *aX*

PARTS LIST

Part	Reference
C1	470nF 100V
C2	470µF 100V
C3	1µF 450V
C4	47N 450
C5	50µF 450
C6	100N 200V
C7	100µF 200V
J1-3	RCA JFem
R1	100K
P1A	100K Log, Dual
R2	1K
R3	1k5
R4	10K 2W
R5	900
R6	270k
R7, R8	47 3W
R9	820 7W
S1	2p3Pos Switch
TP1	H1
TP2	H2
V1	6SL7
V2	6A3

By Joseph D'Appolito

Testing Loudspeakers:
Which Measurements Matter, Part 2

We continue our look at the predictors of quality sound for loudspeakers.

Directional queues come from the first arrival response. We judge arrival direction in well under a millisecond. However, judging what we are hearing takes longer.

To determine spectral balance, the ear-brain combination analyzes the incoming sound typically over a 5 to 30ms interval. This interval is called the Haas fusion zone. Within this interval we are not aware of reflected sounds as separate spatial events. All of the sound appears to come from the direction of the first arrival. Lateral reflections from adjacent walls help extend the soundstage beyond the physical span of the loudspeakers. The comb filtering action of the many early reflections arriving at the listening position with varying phases adds a sense of spaciousness to the sound. (It also argues against the need for phase accuracy in loudspeakers.)

You can see that the perceived timbres of sounds in rooms are the result of temporal processing and spatial averaging of reflected sounds arriving at our ears from many angles. In typical home listening rooms, direct sound and early reflected sounds dominate. Late reflections are greatly attenuated. This is clear from the measurement of RT60s in the range of 0.2 to 0.4 seconds. (Compare this to concert halls where RT60s of 3 to 4 seconds are common.) What we hear is a function of the directional characteristics of the loudspeakers and strong early reflections from the room boundaries.

You can think of the early reflection response as a loudspeaker's in-room response averaged over a period extending out to 30ms after first arrival. But now we have a problem.

The early reflection response is room dependent. A designer cannot predict how the early reflection response will look in any particular room. This will depend on the room size and shape and speaker location. It will also be affected by the room furnishings, any acoustic treatment, and the number of people in the room. However, we can examine a loudspeaker's directional characteristics anechoically. To guarantee that sound arriving at the listening position is affected only by the arriving reflections and not by any off-axis anomalies in the speaker's response, the off-axis response curves should be smooth replicas of the on-axis response with the possible exception of some rolloff at higher frequencies and larger off-axis angles.

POLAR RESPONSE

I have found that the best way to represent a loudspeaker's off-axis response is with a 3D waterfall plot, which is assembled by measuring a speaker's off-axis response at a number of equally spaced angles. The polar data is used to determine the listening window, early reflection, and power responses. DAAS generates a 3D polar waterfall response automatically in conjunction with a computer-controlled turntable. In the polar waterfall option DAAS performs a sequence of measurements. Between each measurement DAAS sends a control signal to the turntable to move a specified number of degrees. The full range of measurement is 180°.

Figure 14 is a polar waterfall plot for my example loudspeaker. To obtain this plot the speaker was rotated in the horizontal plane from 90° left of on-axis to 90° right of the on-axis position in 10° increments. Except for a gentle rolloff at higher frequencies and larger angles, the off-axis curves are excellent replicas of the on-axis curve. You can see that the off-axis response curves are smooth

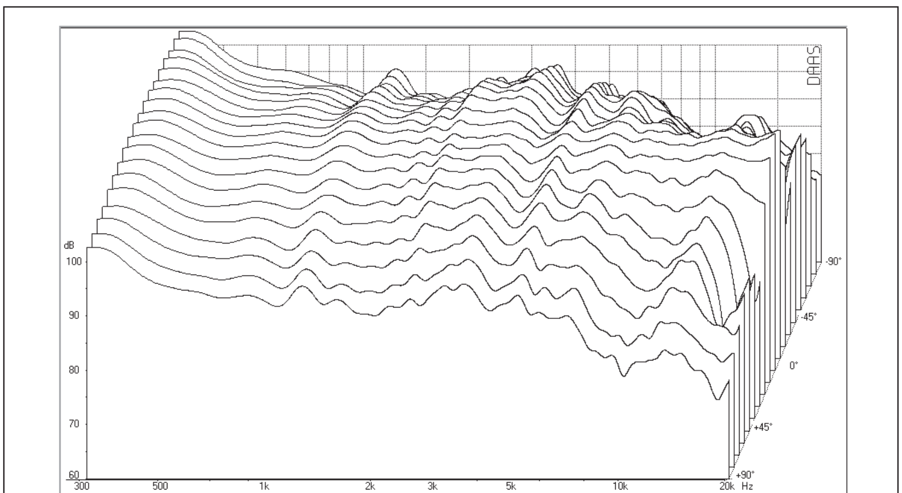

FIGURE 14: Example loudspeaker horizontal polar response.

replicas of the on-axis response with the expected rolloff at higher frequencies and larger off-axis angles due to the narrowing polar response of the tweeter. The high-frequency rolloff produces the desired listening window and early reflection responses of Fig. 1 in part 1.

Figure 14 gives an excellent qualitative view of polar response performance, but reading actual values off the plot is difficult. Using the polar data collected for **Fig. 14**, I have plotted on-axis response and off-axis responses at 30° and 60° in **Fig. 15**. You can see that the 28mm tweeter response falls fairly quickly above 8kHz at 60° off-axis.

Using the polar response data, you can estimate the listening window and early reflection responses. I determined the listening window response by averaging on-axis response with off-axis responses in 5° increments from 25° left to 25° right and between 10° up and 10° down. You can approximate the early reflection performance by averaging all responses in the horizontal plane. The results (shown in **Fig. 16**) agree rather well with the criteria of Fig. 1. The curves shown on this plot appear to have a great deal of ripple, but this is due to the choice of the plot scale. Actually, the on-axis response lies within a ±2dB window above 500Hz.

The power response is obtained by measuring responses at many locations over a spherical volume. This can only be done accurately in an anechoic chamber or, alternatively, in a totally reverberant

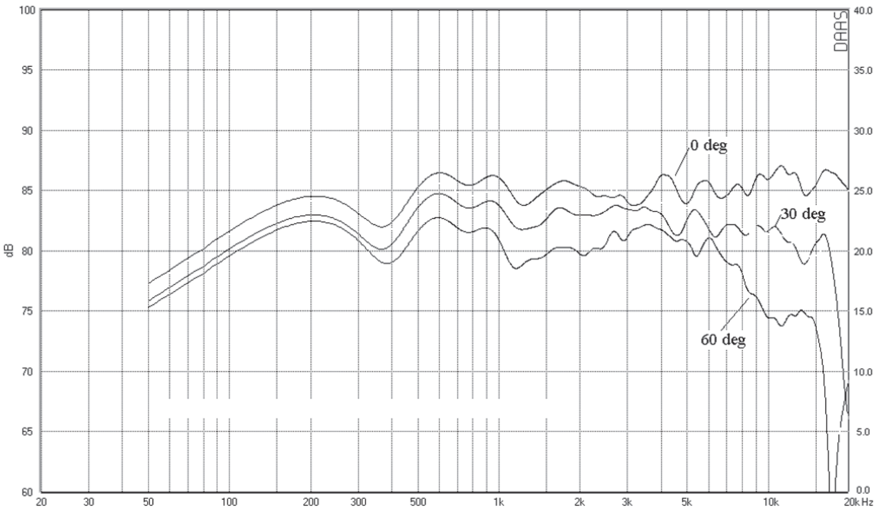

FIGURE 15: Example loudspeaker responses at 0, 30, and 60°.

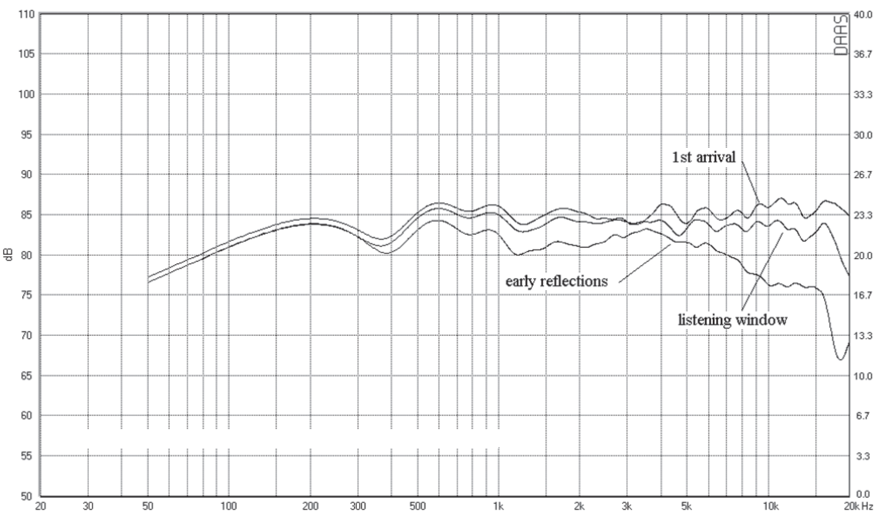

FIGURE 16: First arrival, listening window and early reflection responses.

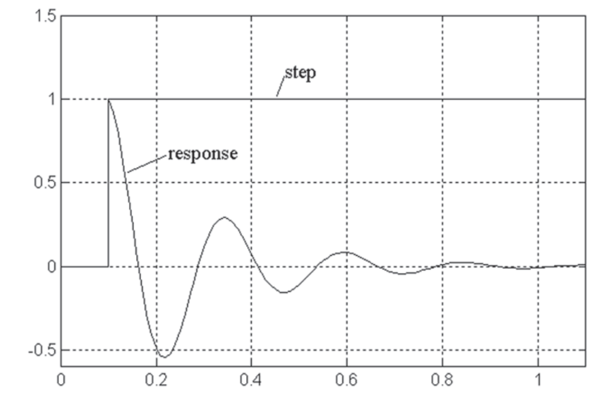

FIGURE 17: Step input and ideal step response.

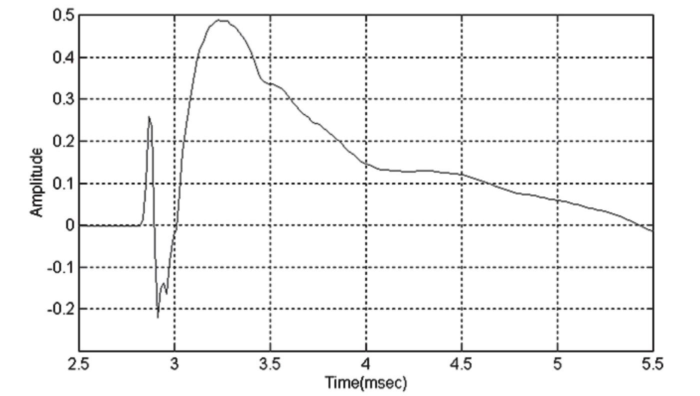

FIGURE 18: Step response of example loudspeaker.

enclosure. Because neither venue is available to me, I cannot show the power response for my example loudspeaker.

STEP RESPONSE

Up to this point we have looked at loudspeaker performance solely in the frequency domain. Let's turn now to the time domain for additional performance insight. We could examine the impulse response in more detail, but it is not easily interpreted. It is dominated by the tweeter response in the first few milliseconds. It doesn't tell us much about the woofer, or the midrange if there is one, because all the low-frequency information is in the impulse response tail, which is at a very low signal level. The step response is a much more useful tool.

The step input is a signal that rises instantaneously from zero to a fixed level. This is basically a DC input starting at time zero. Mathematically, the step response is the time integral of the impulse response.

Figure 17 shows the response of an ideal loudspeaker to a step input. Loudspeakers are high-pass devices that cannot produce a static (i.e., DC) acoustic output. Therefore, the step response must drop below zero for a sufficient time to produce a net output of zero over time. The ideal step response is an exponentially decaying cosine wave oscillating at the fundamental resonant frequency of the loudspeaker.

Figure 18 shows the step response for my example loudspeaker on an expanded time scale. The oscillatory portion of the response is not shown. This plot is actually a combination of two step responses: the initial sharp rise of the tweeter followed by the much slower broader rise of the woofer. This is shown more clearly in **Fig. 19**, where the tweeter and woofer step responses are plotted separately.

What can you tell from these plots? First, you see that both the tweeter and woofer are connected with positive polarity. Both initially rise in the positive direction. Next you see a smooth handoff from the tweeter to the woofer at roughly 3.1msec. This speaks well of the crossover design. Finally, from **Fig. 18** you see that the speaker is not time coherent. Comparing rise times, the woofer is approximately 250µs behind the tweeter.

If you reverse the polarity of the tweeter, you get the step response shown in **Fig. 20**. There is now no longer a smooth transition from the tweeter to the woofer. The frequency response (**Fig. 21**) shows a null in the crossover region of about 12dB due to the tweeter polarity inversion. The response curves shown have been ⅙ octave smoothed. The raw curve shows a notch of greater than 20dB, which is a strong indication that the drivers are in-phase at crossover. I will discuss this condition in more detail in the section on phase response.

Determining driver polarity can be

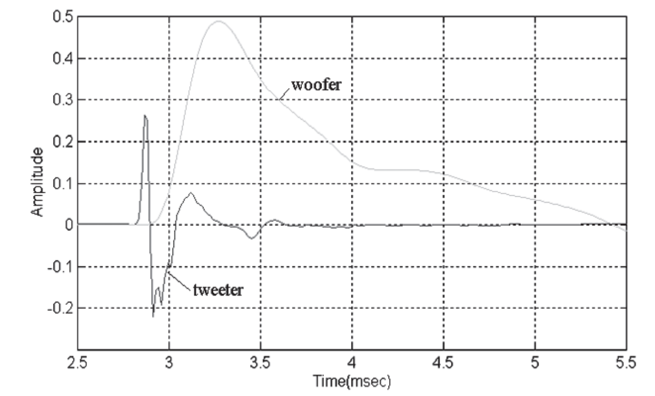

FIGURE 19: Individual driver step responses.

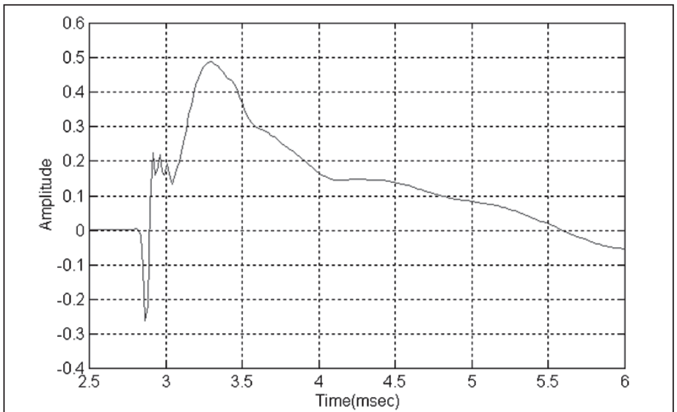

FIGURE 20: Step response-tweeter polarity reversed.

of great value in home theater setups. For example, you may be using a center channel speaker from a different manufacturer than those of the right and left channels. If the center channel tweeter is connected out of phase to get flat frequency response while the left and right channel speakers use in-phase tweeters, you will degrade the imaging of the full system. The effect with woofers can be even more dramatic.

PHASE RESPONSE

Recall that I did not list phase response as one of the predictors of loudspeaker preference. The vast majority of loudspeakers available today are not time coherent and therefore exhibit some degree of phase error. A great deal of research has gone into the subject of phase shift audibility. Papers in the AES and other audio journals are too numerous to reference.

Many researchers employed cascaded all-pass networks in the amplifying chain to introduce several hundred degrees of phase shift over the audible frequency range with no change in frequency magnitude response. The universal conclusion from these efforts is that large degrees of phase shift are not audible when listening to loudspeakers playing typical program material in the semi-reverberant environment of a typical listening room. Trained listeners using earphones have heard differences in sharp transient signals when subjected to very large frequency dependent phase shifts, but this is not the normal listening situation.

There is one possible exception to this conclusion. There is some evidence that large phase errors at low frequencies soften bass drum strikes. Loudspeakers develop large phase shifts near and below their low-end cutoff frequencies. This, in turn, produces group delays on the order of 5 to 15ms in that frequency range. Bass drum fundamentals then lag their upper harmonic components by this amount, which may explain this phenomenon. Countering this effect would require compensating bass amplitude and phase response flat down to below 10Hz or lower.

Notwithstanding the last three paragraphs, there are some things you can learn about a loudspeaker's performance from phase data. Loudspeaker phase data is made up of two components: minimum phase and excess phase. The minimum phase response is related to the dips, peaks, and ripples in frequency magnitude response by a mathematical operation called the Hilbert transform. Any phase shift beyond this minimum phase shift is called excess phase, which is a measure of loudspeaker time dispersion. In particular, excess group delay, the derivative of excess phase with respect to frequency, has the units of time and is a measure of that dispersion.

DAAS measures total phase, computes minimum phase via the Hilbert transform, and then subtracts this from the total phase to get excess phase. DAAS then computes excess group delay from the excess phase. **Figure 22** is a plot of the excess phase response for my example loudspeaker. The plot frequency range has been expanded to cover 500Hz to 20kHz. The excess phase starts out at about 30° and increases rapidly to almost 270° at 6kHz.

More revealing is the excess group delay shown in **Fig. 23**. Referring to the right-hand scale, labeled "Delay/ms," notice that excess group delay is essentially zero at high frequency. As you move down, the frequency axis toward the crossover point excess group delay begins to grow, reaching an asymptotic value of 0.25msec, or 250μsec, below 1000Hz. This confirms the estimate of the tweeter-woofer delay from the step response of **Fig. 18**. Excess group delay is an excellent tool for examining loudspeaker time dispersion.

IMPEDANCE

You can learn a great deal about a loudspeaker from its impedance plot. The impedance magnitude at low frequencies reflects the bass alignment of the speaker. For example, a sealed box speaker will have a single impedance peak at

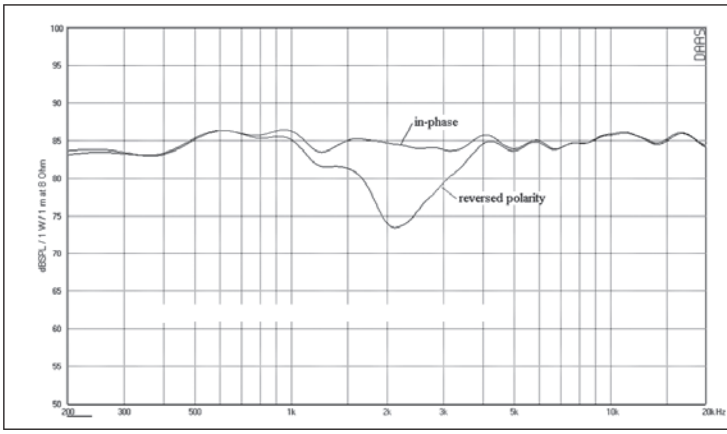

FIGURE 21: Example loudspeaker response with reversed tweeter polarity.

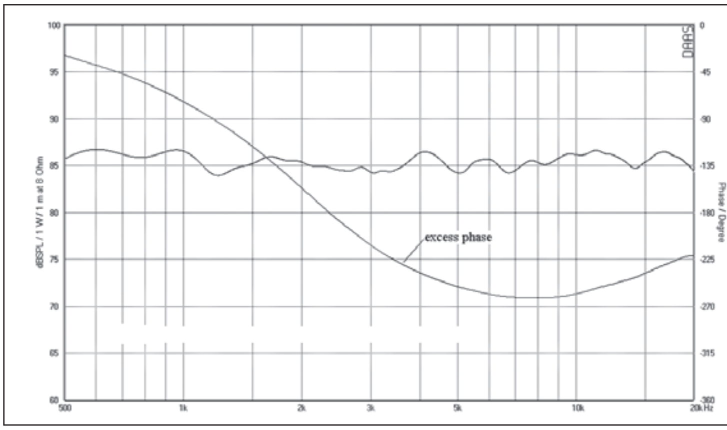

FIGURE 22: Example loudspeaker frequency response and excess phase.

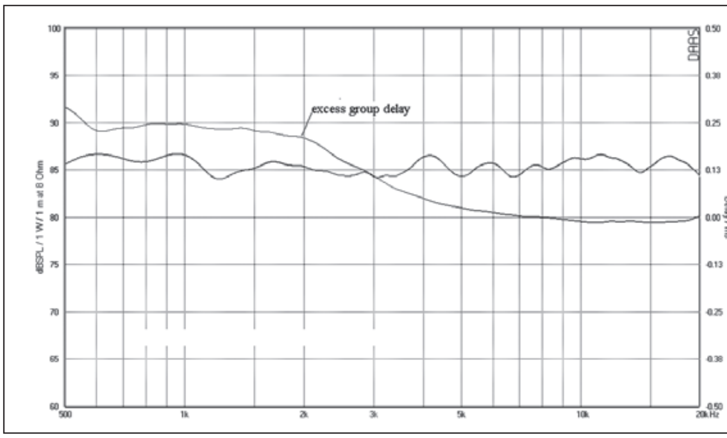

FIGURE 23: Example loudspeaker excess group delay.

its low-frequency resonance. Below this, frequency response will fall off at 12dB/octave.

A vented enclosure will have a double peaked impedance plot at low frequencies. In this case the saddle point between the peaks approximates the vented box tuning frequency. (The woofer voice coil inductance and the crossover circuit may cause a slight shift in the saddle point relative to the actual box frequency.) Below the box frequency the response of a vented speaker will fall at 24dB/octave.

From fine detailed loudspeaker impedance curves you can detect cabinet vibrations and internal resonances such as standing waves. Finally, you can judge how difficult it will be for an amplifier to drive a particular loudspeaker. Very low impedance magnitude values coupled with large phase angles produce large current demands that may be beyond the capability of an amplifier.

Figures 24 and **25** are impedance plots for my example loudspeaker. **Figure 24** covers the full frequency range. The minimum impedance of 6Ω occurs at 3kHz. The phase angle at this point is –22°. The worst phase angle occurs at 2kHz, but the impedance magnitude there is 10Ω. Driving this speaker should not be a problem for any well-designed amplifier.

Figure 24 was generated at a 48kHz sample rate. Reducing the sample rate to 8kHz greatly improves low-frequency resolution. This is shown in **Fig. 25**. From this plot the tuning frequency, f_B, is seen to be about 37Hz. In typical vented box alignments this is approximately the –6dB response level.

The impedance plot is also a diagnostic tool. **Figure 26** is the impedance plot for a fairly well-regarded two-way tower loudspeaker. The impedance is shown on the same expanded scale as that of **Fig. 25**. Notice the glitch in both the magnitude and phase plots at 165Hz.

There are three possible causes for this wrinkle in the impedance plot: port tube resonance, cabinet vibration, or a standing wave in the enclosure. Port tubes can develop organ pipe resonances. The port tube for this speaker is only about 15cm long, so I doubt port tube resonance is the cause because any organ pipe resonance would be much higher in frequency, but you can prove this rather simply. **Figure 27** is an impedance plot of the

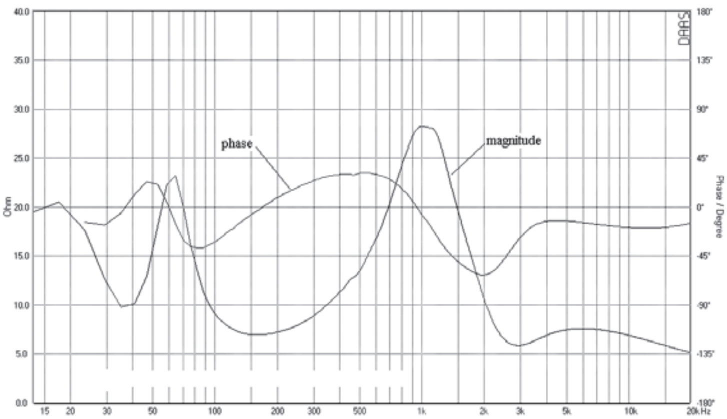

FIGURE 24: Example loudspeaker impedance magnitude and phase.

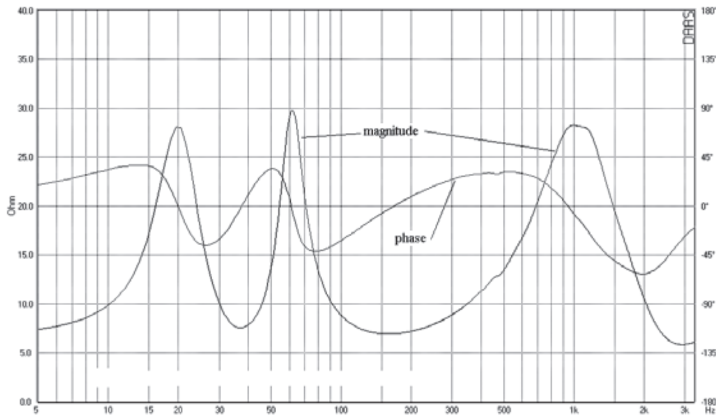

FIGURE 25: Example loudspeaker impedance on an expanded scale.

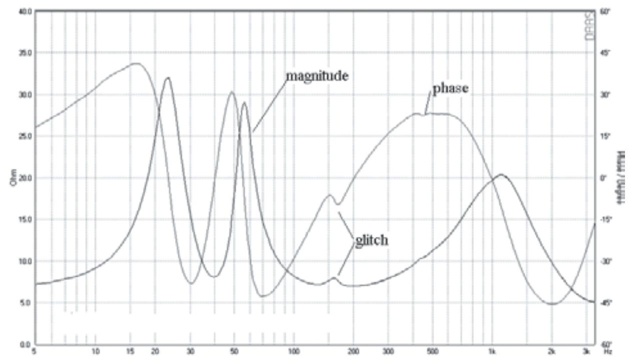

FIGURE 26: Tower loudspeaker impedance plot.

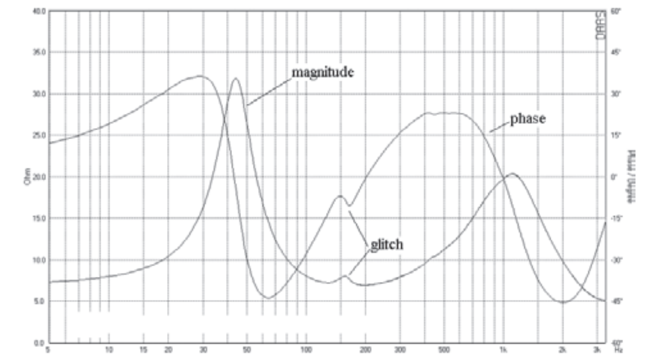

FIGURE 27: Tower speaker impedance with vent plugged.

speaker with the port tube plugged with a large block of polyfoam. Now there is the single peak characteristic of a sealed box alignment. But the glitch at 165Hz is still there and unchanged. So port resonance is not the problem.

Next side panel acceleration was measured with a Measurement Specialties ACH01 accelerometer driving the speaker with a 5V swept sine wave. The resulting acceleration spectrum is plotted in **Fig. 28**. Acceleration levels were so low I thought most of the measurement might just be accelerometer noise. So I measured the self-noise of the accelerometer, which is also plotted in **Fig. 28**.

Up to 200Hz acceleration levels are just slightly above the noise level. Acceleration peaks in the 400 to 500Hz range at about –60dBV, which translates into an acceleration level of about 0.14g. This may seem like a high level, but at 500Hz this amounts to a panel displacement of microns. More important, there is no acceleration spike at 165Hz. So panel vibration is not the problem.

This leaves the possibility of a standing wave. **Figure 29** plots the low-frequency response of the tower speaker. This plot was obtained using the previously described feature in DAAS for combining near-field woofer and port outputs. This plot is valid up to about 300Hz. Notice the small dip at 165Hz, which represents the power taken from the woofer output to sustain a standing wave within the enclosure.

Now I could have guessed the problem was a standing wave from the beginning if I had first described the physical appearance of the speaker, but I wanted to highlight the analytical tools available to examine this problem. The tower internal height is 102cm. The woofer is mounted at the top of the front baffle. So you have a closed pipe excited at one end. Assuming a sound velocity of 343m/sec, this calculates out to a standing wave at 168Hz. Given a small amount of filling material which may slow sound speed in the enclosure, 165Hz seems right on.

EFFICIENCY AND SENSITIVITY

A loudspeaker's efficiency tells you how much acoustic power and sound pressure level a loudspeaker can produce for each electrical watt of input power. It is specified in terms of the sound pressure level generated with 1W input at a distance of 1m, i.e. dBspl/1W/1m. Because loudspeaker impedance varies widely over frequency both in magnitude and phase, it is difficult to determine the true input power to a loudspeaker. To get around this problem a constant resistance is assumed for the loudspeaker impedance. Typically a value of either 4 or 8Ω is used.

The assumption of a constant resistive impedance in the efficiency measurement means that frequency response tests which are nominally made with constant input power are actually made with a constant input voltage. Modern solid-state amplifiers are essentially constant voltage sources. As long as output stage current limits are not reached, these amplifiers will provide whatever current is required to meet the demands of the constant voltage frequency sweep. For this reason it is now common practice to specify loudspeaker performance in terms of voltage sensitivity, S_0, which has the units of dBspl/2.83V/1m. 2.83V represents the voltage that will produce 1W of power dissipation in an 8Ω resistor.

I have measured hundreds of loudspeakers for sensitivity. In my tests values have ranged from 84 to 91dB. All other things being equal, the higher the sensitivity the better. Unfortunately, all other things are rarely equal. Speakers with the smoothest frequency response are often not the most sensitive.

As part of the frequency response measurement, DAAS calculates the distance

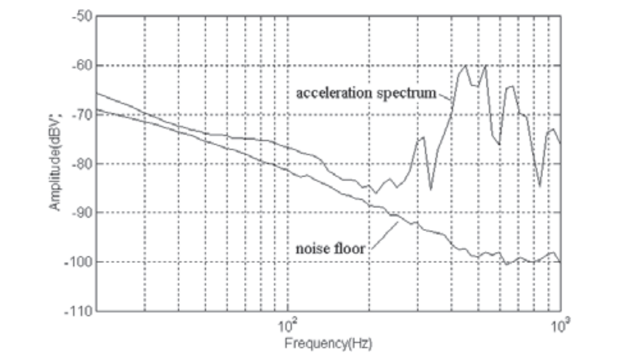

FIGURE 28: Side panel acceleration spectrum.

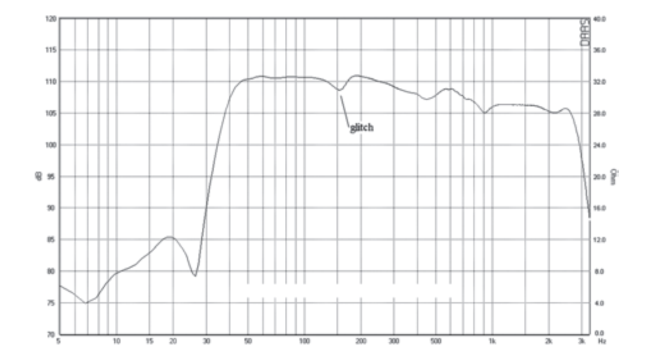

FIGURE 29: Tower speaker low-end frequency response.

from the microphone to the loudspeaker under test and automatically references the measurement to 1m. I then calculate sensitivity as the mean SPL over the 500Hz to 2kHz band. Looking at Fig. 3 in part 1, the sensitivity of my example speaker is 85dBspl/2.83V/1m.

DISTORTION

First, you must distinguish between linear and nonlinear distortion. If a loudspeaker is linear, doubling an electrical input signal will exactly double its acoustic output. If two frequencies, f_1 and f_2, are input to a linear loudspeaker, only those two frequencies and no other frequencies will appear in the output. Any departure from flat frequency response will distort a signal. This is linear amplitude distortion. The relative magnitudes of f_1 and f_2 may change, but no additional frequencies are produced. On the other hand, nonlinear distortions such as harmonic and intermodulation distortion produce signal components not in the original program material.

I have not found a quantitative or qualitative relationship between the various distortion types you can easily measure and loudspeaker preference. The audibility of nonlinear distortion is a complicated issue. It is relatively easy to detect a few percent distortion in simple signals such as a pair of sine waves. However, large levels of distortion can be tolerated in complex program material such as rock 'n' roll music. In my experience, the maximum sound pressure level a speaker can generate is dictated by the level of distortion the listener will tolerate.

Distortion measurements do not directly predict how a speaker will sound, rather they help us judge driver linearity and by implication driver quality. DAAS implements tests for harmonic and intermodulation distortion. Although I will show some harmonic distortion test results, I believe that intermodulation distortion tests are more revealing of loudspeaker performance. We can tolerate relatively high levels of harmonic distortion in program material because, as their name implies, the spurious components added to the program are harmonically related to the original program.

Intermodulation distortion (IMD) produces output frequencies that are not harmonically related to the input. These frequencies are much more audible and annoying than harmonic distortion. In one kind of IMD test two frequencies are input to the speaker. Let the symbols f_1 and f_2 represent the two frequencies used in the test. Then a 2nd-order nonlinearity will produce intermods at frequencies of $f_1 \pm f_2$. A 3rd-order nonlinearity generates intermods at $2f_1 \pm f_2$ and $f_1 \pm 2f_2$.

I ran a harmonic distortion test on my example loudspeaker. Average level was set at 90dB/1m. **Figure 30** plots the 3rd and 5th harmonic distortion levels out to 8kHz. The test consists of a sequence of 50 distinct frequencies from 100Hz to 8kHz. Only the odd-order distortion products are shown because these are known to be most objectionable.

Distortion components average about 0.32% up to 300Hz and drop to below 0.1% beyond that point. The one outlier of 1% at 180Hz is a false reading caused by a panel vibration in my lab interfering with the speaker output at the mike location. Apparently the resonant frequency of this panel aligns almost perfectly with test frequency of 180Hz.

Figure 31 shows the results for the same test run on the PA speaker at an average level of 87dB/1m. (Distortion was excessively high at 90dB/1m.) Notice that tweeter 3rd harmonic distortion rises to 1% above 3kHz. The 5th–order distortion averages 0.32% above 300Hz. It is clear that the drivers used in the PA speaker are of poorer quality than those used in my example speaker.

Next I examined IMD in the woofers. **Figure 32** shows results for the two-tone

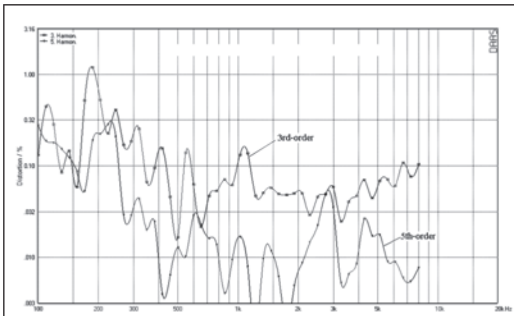

FIGURE 30: Example loudspeaker odd-order harmonic distortion.

IMD test run on my example speaker at the 90dBspl level. The two frequencies of 300 and 1300Hz were picked to exercise the woofer. There is a 2nd-harmonic of the 300Hz tone at 600Hz, but it is down 60dB from 300Hz. The only significant IM product is a 2nd-order one at 1600Hz. It is down 56dB from the full output. The same test was then run on the PA speaker. Examining **Fig. 33**, you can see significant IM products at 700, 1000, 1600, and 2200Hz.

DYNAMICS

How often have you turned up the volume only to feel that the music is not getting louder? The sound stage seems to collapse, transients dull, and the sound becomes congested and lifeless. You are experiencing short-term dynamic compression. You have exceeded the SPL capability of your loudspeaker. When listening to classical music, short-term transients may exceed the average sound level by 12 to 20dB. If the program material increases by 12dB, but your speaker output only increases by 10dB, you are experiencing dynamic compression.

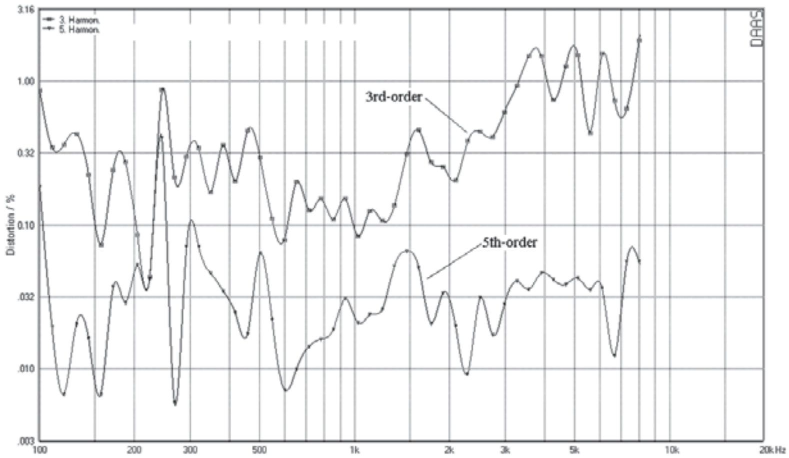

FIGURE 31: PA speaker odd-order harmonic distortion.

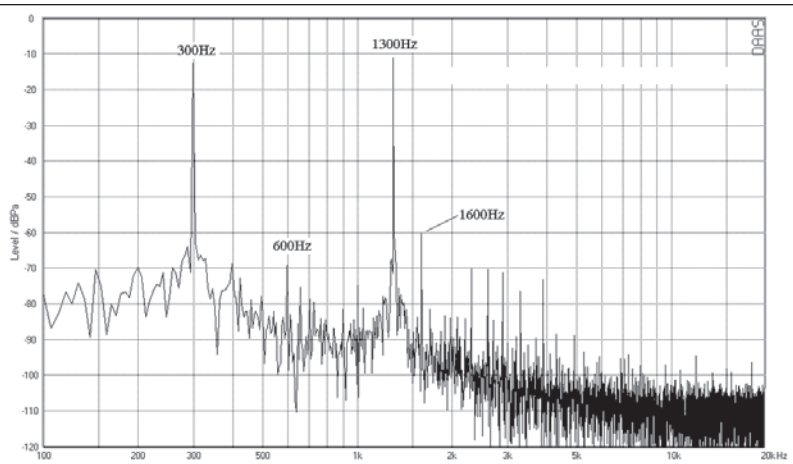

FIGURE 32: Example woofer IM distortion.

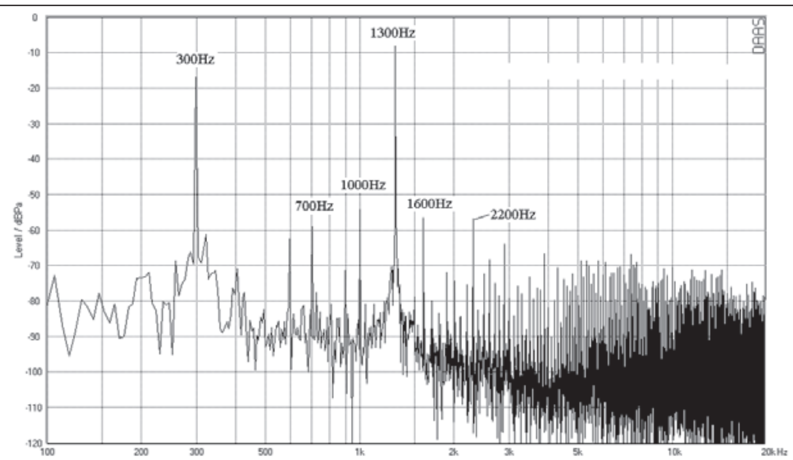

FIGURE 33: PA speaker woofer IMD.

Short-term dynamic compression should not be confused with power compression. In sound reinforcement applications such as rock concerts, the average power level fed to the loudspeakers is quite high. Under this condition driver voice coils heat up. The coil resistance increases, reducing the driver sensitivity. This is power compression.

In typical home listening environments, average power levels are only a few watts at most. Voice coil heating is not much of a problem. In this case the compression arises out of some nonlinear behavior of the driver compliance or magnetic field distribution such that the driver cone excursion does not keep up with the input voltage demand.

DAAS has many interesting signals in its signal library that can be used to test loudspeaker dynamic response. One signal consists of a set of eight sine waves spread out over an interval from 500Hz to 2.5kHz. The spectrum of this signal is shown in **Fig. 34**. This signal can be played as a single event and the resulting SPL measured. Then the signal can be quickly increased by several dB and played again.

When playing classical music, average SPL levels are typically in the low 80s. Using my example loudspeaker, the test signal was first played for 170msec at a level of 82dB SPL. The signal was increased by 15dB and played again. The output rose to exactly 97dB. Looks like my example loudspeaker has good dynamics (**Fig. 35**).

SUMMARY

We have seen that the single best predictor of loudspeaker listener preference is frequency response. There are four elements to frequency response: on-axis response, listening window response, early reflection response and, finally, power response. The last two require that you also examine polar response. Resonances are the principle causes of objectionable sound. Strong resonances are often obvious in the frequency response plots. However, the CSD and PCSD provide us with more detail and often reveal delayed resonances not obvious in the frequency response alone.

Turning to the time domain, the step response gives us qualitative information on driver polarity, time dispersion, and driver integration. Phase response is not a strong indicator of speaker quality, but we can glean more detailed information on speaker time dispersion from the excess group delay plot. Impedance data can be used to detect cabinet vibrations and internal resonances such as standing waves. We can also judge how difficult it will be for an amplifier to drive a particular loudspeaker. Very low impedance magnitude values coupled with large phase angles produce large current demands that may be beyond the capability of an amplifier.

Unless distortion levels are very high, harmonic and IM distortions are not strong predictors of listener preference, but they are useful in assessing driver quality and can explain why speakers sound bad when played at high volume levels. The dynamic capability of a loudspeaker is a very strong predictor of its ability to produce lifelike sound. Finally, the measurements discussed here are not only useful in evaluating existing designs, but they can also be used by loudspeaker engineers as design goals.

There is one caveat in all these results. The discussions here have been limited to conventional, forward-firing dynamic loudspeaker systems. Large panel loudspeakers and line arrays present vastly different measurement challenges. In the home listening environment, you will invariably be in the near field of these speaker types. Response will vary widely with listener position in height and distance to the speaker. Defining a single response axis that characterizes one of these speakers is difficult. Also, polar response will differ substantially from conventional speakers. *aX*

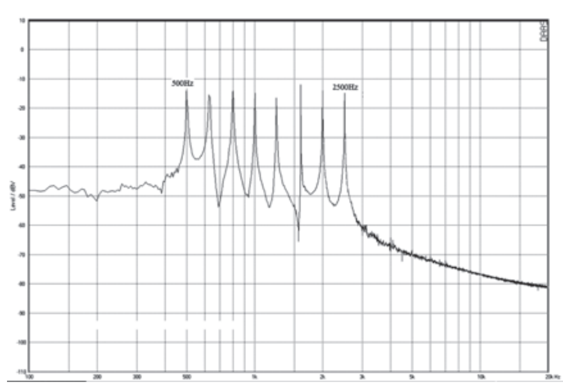

FIGURE 34: Test signal spectrum dynamic response test.

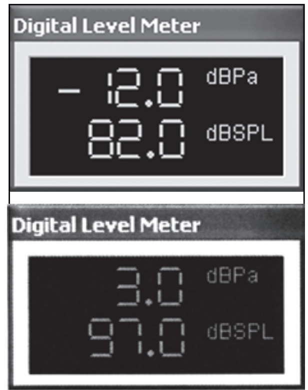

FIGURE 35: Sample SPL levels.

A note on testing: All measurements used in this article were made with either the DAAS4usb or the DAAS4pro192 PC-controlled acoustic data acquisition and analysis systems. Acoustic data was measured with either a calibrated Earthworks MD30 microphone or ACO Pacific 7012 ½″ laboratory grade condenser microphone and a custom designed wide-band, low-noise preamp. Cabinet vibration was measured with a Measurement Specialties ACH01 accelerometer. Polar response tests were performed with a computer-controlled OUTLINE turntable on loan from the Old Colony Division of the Audio Amateur Corporation. ■

SPECIAL OFFER

This two-part article by noted audio expert Joseph D'Appolito is available for free on CD when you purchase his book, *Testing Loudspeakers*. His definitive work on how to reliably test loudspeakers is available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, 603-924-9464, FAX 603-924-9467, e-mail custserv@audioXpress.com.

By Ed Simon

An Isolation Transformer, Pt. 3

Parts 1 and 2 (August and September *aX*) covered why you need to clean up a power line and two ways to do it. Here are some additional methods.

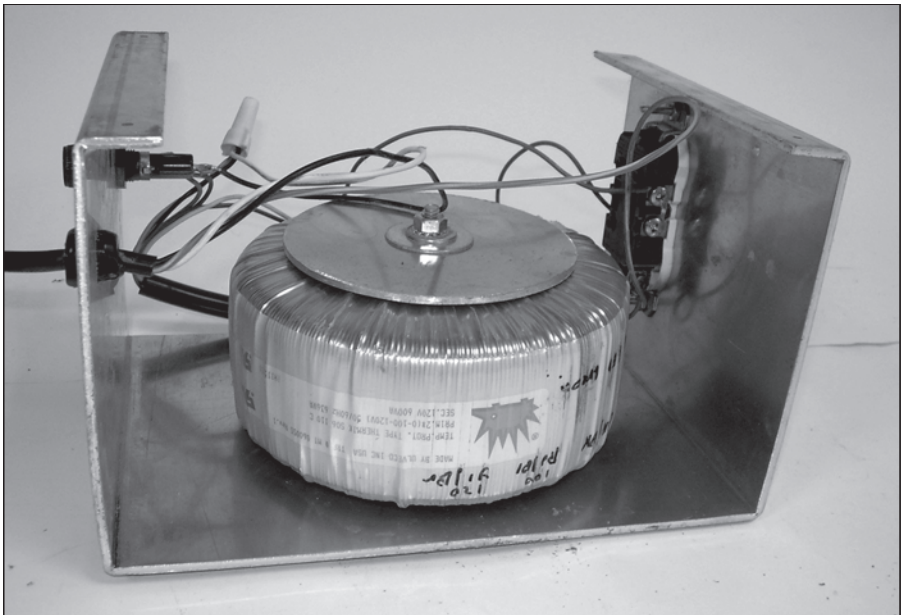

PHOTO 1: Isolation transformer.

One of the most common methods to clean up a powerline is to use an isolation transformer. Unlike an audio transformer, which is designed to have a wide bandwidth, this special kind of transformer is designed for a narrow frequency response.

DESIGN DECISIONS

Transformer design is no longer considered a hot field and is not often offered in schools today. So, of course, my education included a semester of magnetics. I can actually design a transformer that does not catch fire. (Smoke does not count!)

The first area of concern is designing the magnetic core. There are two popular structures: the toroid or continuous tape wound core and the E-I core. The toroid is becoming more popular today because the costs of making them have decreased. There is a bit of debate as to which style is preferred for audio power supplies. I generally find the values I need are offered in only one style, solving that design decision!

Transformers use a coil(s) of wire to produce a magnetic field and then another coil(s) to turn it back into electric current. The ratio of the turns from the primary to the secondary determines the relationship of the input voltage to the output. There is some loss in the process, but it is usually not of much concern in small transformers.

To confine and allow a greater magnetic field, a core is used. The problem with a core is that at some point the material saturates, just as the toy telegraph did. There are other issues, such as the magnetic field increase is not linear current over most of the useful field range, the core can act as a shorted turn and eat up energy, there is extra inductance in just winding the coils, and, of course, there is also capacitance in each coil and between them.

If a good transformer designer works at it, he/she can adjust the coil windings for a wide or narrow bandwidth. The winding designs can minimize or maximize the inductance and capacitance of the windings.

One of the important considerations for an isolation transformer is the inherent capacitance between the primary and secondary windings of the transformer. There are several techniques to minimize this. The most popular is to wind the primary on the coil form first, place a copper foil shield that is insulated on one side to prevent it from acting as a shorted turn, and then to wind on the secondary. A more recent method is to use a plastic bobbin that winds the coils side by side to minimize the coupling. I have seen some claims that just using a toroid core minimizes coupling.

The copper foil Faraday shield is found in the transformers of most professional audio gear. However, I rarely find these types of transformers offered for general use.

In addition to placing a shield, with careful design of the windings and core material, the bandwidth of the transformer may be reduced to prevent high-frequency noise from going through. If, for some strange reason, you need to know a bit more about the design of small transformers for both power and audio use, this topic is briefly covered in the *Radiotron Designer's Handbook* available from Old Colony on CD.

CLEANING THE LINE

I could build an isolation transformer. The easiest way is to take an old transformer apart. If you decide to do this completely, be sure to count the number of turns in the primary winding. You will need that number of turns, no matter what size wire you use, to avoid saturating the core. Of course, in a few cases you can save the primary winding and just remove the secondary, making sure to count those turns.

Based on either the primary or secondary windings, you can then figure out how many turns per volt you need to get your desired secondary voltage. Of course, add 10% for winding resistance and core loss. Size your wire for the current you need. The original core size will determine the total volt-amperes the core can handle. You just can't rewind a 24V 1A transformer for 120V at 5A. 200mA is about all you will get.

If this sounds too much like work, it is. I will sometimes modify a transformer by adding an extra turn or two for an additional secondary. You can use this as a high current winding or place it in series with one of the other windings to adjust the voltage.

I have also wound application-specific transformers for prototypes using a commercial core. For this project I will buy a ready-made transformer, and there are several to choose from.

This transformer can have a small voltage step-up to correct for the flattening of the waveform and the losses in the pre filters and cable. It should be designed as an isolation transformer, meet the applicable codes and standards, and even be able to do a few more tricks. **Photo 1** is my plain vanilla isolation transformer that lives on my test bench.

To take the idea of AC line improvement a step further, you can use balanced power. Normal AC supplied for residential use actually is balanced power! The normal feed is two lines called the hot wires and a center tap called the neutral. The neutral wire is grounded, but it is not used for the safety ground. That is a separate wire connected to the neutral and building ground at one point only in the properly wired home.

When you use a single line to the center tap or neutral, you get your typical 120V AC or half of the balanced 240V that is really provided. The National Electrical Code recognizes that when a line is balanced, noise rejection is improved for audio purposes.

With a center-tapped isolation transformer, you can take the single unbalanced AC line and balance it. The advantage of this system is an almost total elimination of hum from musical instrument grade sound equipment. It won't hurt the better stuff either. I show this circuit (**Fig. 1**) with a Plitron transformer made for the purpose. The other part numbers are from Digi-Key.

One big concern with balanced power is that a simple switch on the output will turn off the equipment but may not leave it safe! To fix that, you need a double-pole switch. The other safety issue is to be sure to build this in a metal box. This transformer isolated line cleaner costs a bit more than the resonant filter version.

This circuit (**Fig. 1**) can handle 20A. That means that the 3AG style fuse that was fine for the earlier circuit may be too small for this version. A "Midget" fuse is the next size up and will work well here. Check to see whether your outlets are rated for 20A before you build this version. A 20A outlet will take a 15A plug, but a 15A outlet will not take a 20A plug. A 20A outlet has a T-shaped slot on the hot side jack. A 20A plug has one blade perpendicular to the other.

I show this circuit with a 15A fuse because that is what most outlets provide. If you choose to use this for the full 20A rating of the transformer, you will need a proper plug, fuse, and a higher current input switch.

The case can come from the electrical supply house such as in the earlier version. However, because this circuit will power more devices, you may want to cheat a bit on mounting all the AC outlets. Buy a surface quad box or two with plates and a few box connectors with bushings. Now one or two 5/8″ holes will do for all of the wiring.

You now have met a reasonable set of engineering goals. You are getting lots of current, restored the peak voltage, and cleaned up much of the line noise. A good question is "Can you do better?"

AC POWER SUPPLY

Yes, you can. So far everything you have

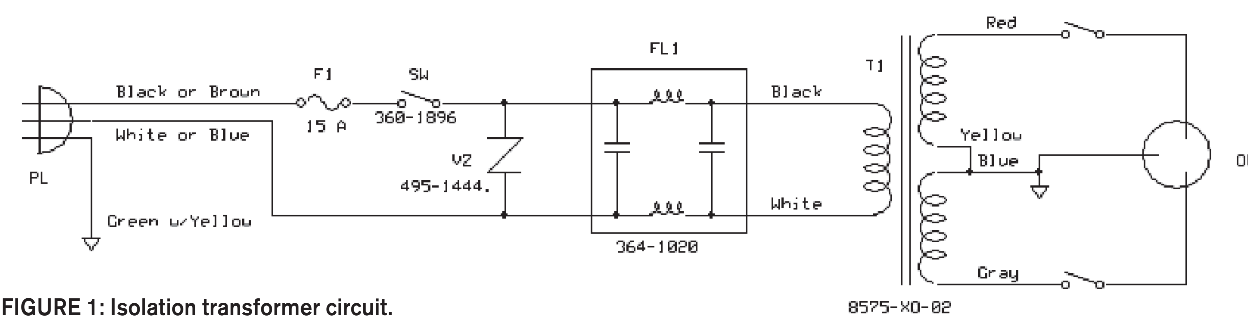

FIGURE 1: Isolation transformer circuit.

done is with passive devices. If you use active devices, you can make the power line appear to have almost no resistance or, in other words, make the line stiffer. I have already mentioned how as the power supplies draw current the voltage drops. When your power amplifier charges the filter capacitors, your AC peak line voltage drops. You can reduce this compression by using a regulated AC power supply.

The simplest method of achieving a regulated AC supply is to hook up a 60Hz oscillator to a very large power amp (one that uses feedback to stabilize its output!) and use it to provide AC mains. You can then set the voltage to whatever you need. Some of the older vacuum tube gear was designed for a line voltage of only 115V, while some modern gear likes 126V. Some gear works better when the supply frequency goes up to 90Hz.

This may seem like a costly and inefficient way to achieve power. It is. For safety reasons, you should start with a power transformer to get the line isolation. Then you will need some large rectifiers. It is hard to feed capacitor steroids, so you will need some really giant ones.

The output transistors are doable even though you will need at least a dozen. It is the heatsink that you might want to get three estimates for. Ed Dell supports the principle that amateurs can build professional-quality gear both in appearance and performance and save a bundle doing it. Besides, it is fun.

However, if you wish to try the high-power regulated AC approach, I suggest you purchase a used Crown MA5000-VZ amplifier, which can easily deliver 20A at 120V if you can feed it properly. These amps were the mainstay of professional sound reinforcement for many years and are now being replaced by even bigger Crown amps. So as an amateur you have a rare chance to take advantage of the professionals' need for "new, bigger, and shinier." Of course, if price is no object you could buy the brand new version, the MA5002VZ.

Just put the amplifier in the mono bridging mode and connect your outlets to the two red terminals. Don't forget to fuse this line! Crown would like you to put a very large capacitor resistor network in series with the line to keep the amp happy under worst-case inductive failures. That is your call. Be sure to check the voltage and frequency before you connect your gear! You may even wish to keep a monitor on the output all the time. If you have a problem, check the VZ switches hidden under the front air filter.

Of course, you can still build the oscillator to task the MA5000VZ.

You could filter the incoming 60Hz line and use that for the reference. One difficulty is that even after cleaning up the noise and distortion the voltage is not constant. You also would need to add some sort of variable gain amplifier (VGA) after the filter. The VGA is adjusted by comparing a reference voltage to the actual output to make sure that as the incoming AC line varies due to changing loads and the whims of the electric company, the supply does not. This could be made to work, but seems complicated.

You could build a classic Wien-bridge oscillator, but this introduces the problem of settling time, and the output voltage is still not regulated.

If you thought to use a phase locked loop, that also would work. A simple oscillator is compared to the zero crossings of the AC line and sped up or slowed down to match. Some sort of lock out would be needed while the oscillator locked on to prevent really strange things from happening. Also, a level control would again be needed for the output reference.

Another choice is a microprocessor feeding a D-A converter to produce a sine wave. Very do-able. Although the hardware would include just a few chips, much software would be required. . . and more work than I like to do.

Of course, it is even simpler to just have a microprocessor output line drive an RC low-pass filter, but I prefer to use a weirder type of circuit—a square-wave oscillator feeding a transversal filter! Sounds like a digital analog circuit! Actually, it is a discrete time continuous voltage device.

A 555C timer is the clock for the entire system. I chose a 64 stage filter. Because my filter works on essentially half of the sine wave, the clock needs to be 128 times faster than the output signal. This is 7680Hz for a 60Hz output. I have added a variable resistor so that you can try output frequencies from 50 to 90Hz.

The clock signal tells my discrete time delay line when to shift the data, which is just the clock signal divided by 128. Either it is high (5V) or low (0V). The input signal is a square wave whose level is very stable. I use a common inexpensive counter IC—type CD4020—to give me my data signal.

This square wave feeds the 64-step delay line, which is made up of eight 74HC164N serial-in parallel-out shift registers. CMOS process logic tends to have very nicely controlled output impedances for this use while using almost no power.

FILTER OPERATION

At first, all of the outputs are low, giving zero volts out. Then at the next clock cycle the first output goes high to 5V. The rest stay at zero. The first resistor causes the output to move up a bit. Then at the next clock cycle outputs 1 and 2 are high, giving a greater output. This continues until all the outputs are high and the peak of the sine wave is reached. The output 1 goes low, starting the downward journey.

Selecting the right resistor values gives me a good approximation of a sine wave. Note that the voltage reference is the actual power supply voltage. Here I use common 7805 type regulators, which, by themselves, are actually not quite as accurate as they could be. By using a regulated 12V source to feed the regulator, the double regulation seems to be adequate. The actual regulation seems to vary depending on who makes the regulator chip.

If you use all CMOS logic devices, the current draw is so small most of the current is used by the regulator chip. For the 12V regulators I use a classic

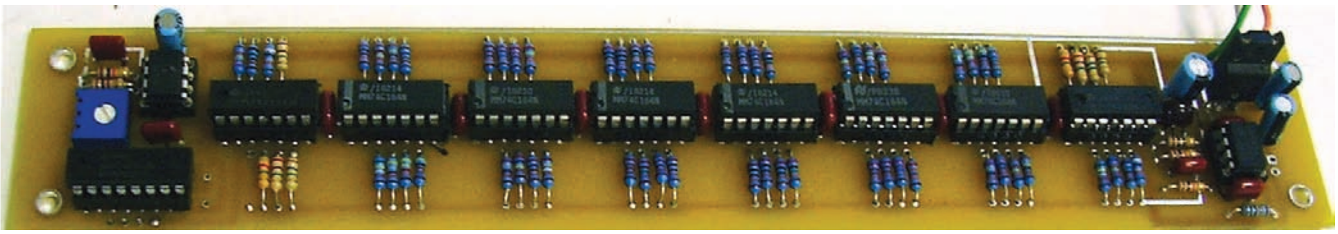

PHOTO 2: Circuit board.

shunt regulator in the simplest form. A resistor and zener diode give me more than adequate regulation, and the cost is minimal.

The classic transversal filter works by multiplying each output by a carefully calculated constant to produce a filter. In this case I want a filter that will give me a sine wave.

I am able to calculate what the constant is for each filter tap to get the desired waveform. If you wanted a universal filter, you would use an analog multiplier for each tap. I just use a resistor because the tap values never change. I only want one filter result!

A large resistor is the same as multiplying by a very small number and a small resistor approaches multiplying by 1. Just be sure to include the resistance internal to the shift register chip when picking resistor values. I find keeping the values above 10kΩ reduces this effect.

Because the resistor values are within only 1%, it seems reasonable to use around 100 steps for the filter. 128 is a good binary choice and easiest to implement.

The results of all of these multiples are summed to form the correct value (voltage) by a single op amp. The result should be a sine wave of 60Hz made up

PARTS LIST

Part #	Value
R1	1G/Open
R2, 64	8.2M
R3, 63	2.7M
R4, 62	1.6M
R5, 61	1.1M
R6, 60	909K
R7, 59	750K
R8, 58	634K
R9, 57	549K
R10, 56	487K
R11, 55	442K
R12, 54	402K
R13, 53	365K
R14, 52	340K
R15, 51	324K
R16, 50	301K
R17, 49	287K
R18, 48	274K
R19, 47	261K
R20, 46	249K
R21, 45	243K
R22, 44	237K
R23, 43	226K
R24, 25, 41, 42	221K
R26, 40	215K
R27, 28, 38, 39	210K
R29, 30, 36, 37	205K
R31, 32, 33, 34, 35	200K

of 128 segments.

The distortion of this waveform will be mostly determined by the lack of accuracy in my multiplier. In other words, the resistors' tolerance determines the distortion. Using 1% resistors for the values under 1MΩ and paralleling them reduces the individual contribution to the total error. I would expect this distortion to be less than .25% for the values used here. A simple low-pass filter in the feedback of the op amp set at about 600Hz should ensure this distortion can be lower.

Figure 2 is the actual output of this oscillator. I am not sure whether the digital artifacts are due to the signal or the digital scopes' resolution.

The same filter taps are used for both the positive- and negative-going excursions of the waveform; as a result, the even-order harmonics are very effectively cancelled. As shown here (**Fig. 3**), the second-order product is almost 95dB down. The highest distortion product is the third harmonic, which is down by 55dB. This is a distortion of .18%. There is also a bit of clock noise that comes through, but it is even lower. You could add a filter to lower this, but I wouldn't bother.

The filter on the output op amp (**Fig. 4**) causes the voltage to drop by 1% at 90Hz. I used a golden oldie for the op amp: an LM301. I chose it because it uses external compensation, which allows me to reduce the bandwidth to decrease noise and distortion.

You could place this circuit in a small box, or even on a small circuit card and insert it into the space provided in an MA5000VZ amplifier. The current draw is low enough that it could be powered by the amp's internal power supply. I built it on a small PC card.

No power supply is shown because the amplifier usually can provide the power. It requires ±15 to 24V at 15mA.

You can, of course, also use this basic design for a wide range swept oscillator. With different tap values, it can produce other waveforms. A cute modification is to use the shift registers as a pseudorandom counter and adjust the taps for a pink noise filter.

If using a monstrous amplifier and oscillator seems like overkill, it is. Perhaps as a more modest approach you can use a smaller amp with a step-up transformer to provide a clean AC source but at a lower total power level.

I will look at that option next month in part 4. *aX*

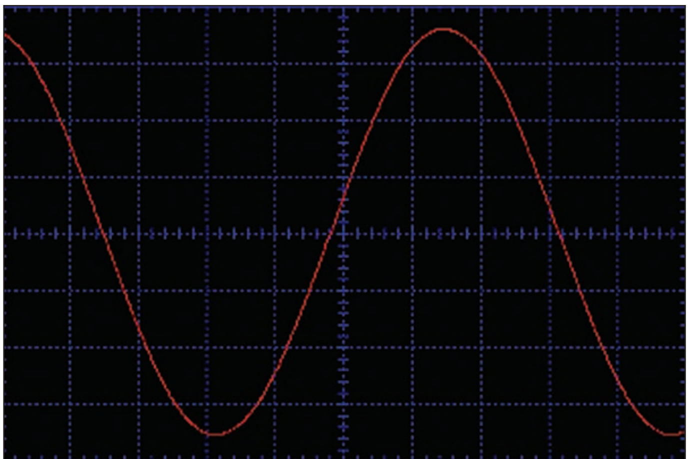

FIGURE 2: Output of the oscillator.

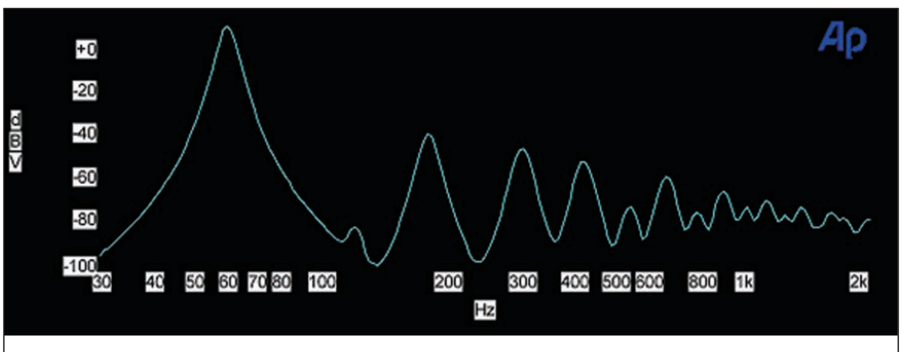

FIGURE 3: Noise and distortion response.

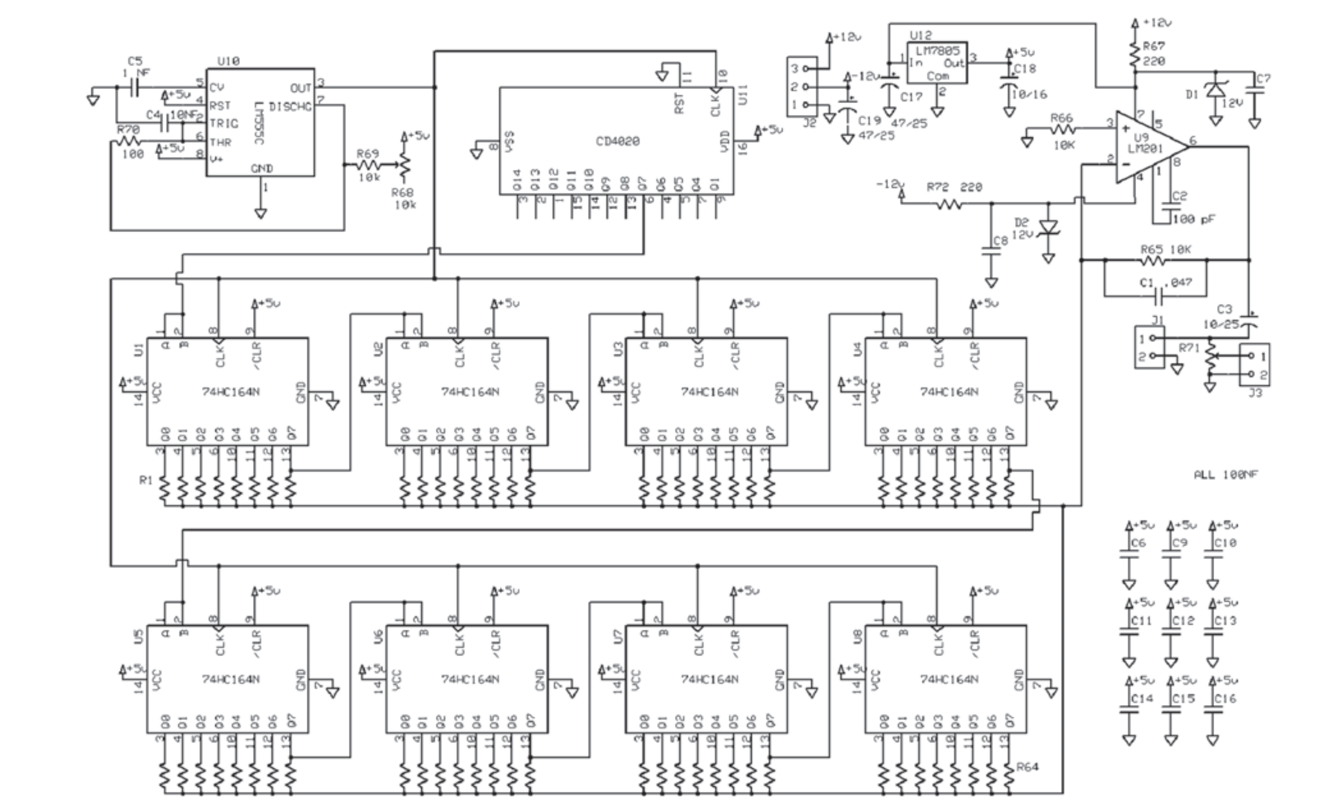

FIGURE 4: Filter layout.

CORRECTION

I spotted a mistake in equation (VII) in my article "Choosing Cathode Bypass Capacitors" (*aX* 8/08), for which I alone am responsible. There should have been a closing bracket after ra in the denominator. It should be as set out in equation (VIII). Because equation (VIII) is right, no one should be too confused, however.

$$\omega R_k C_k = \sqrt{1 + \frac{R_k(\mu + 1)}{2(Ra + ra) + \frac{1}{2}Rk(\mu + 1)}}$$

D. Ivan James

MONSTER MONITOR

If you prefer a large screen for your work such as drafting, drawing, circuit schematics or large-scale calculations, you might welcome our David J. Weinberg's resource search for such a satisfactory upgrade from a smaller unit to what he refers to as "a tame monster." Some of us prefer dual monitors as a solution to video real estate which is fairly easy these days since video cards now routinely offer two video outlets. For the more adventurous go to www.multimediamanufacturer.com, or refer to the article in the Sept./Oct. issue of *Multi Media Manufacturer*, to benefit from David's advice.—ETD

DIY FESTIVAL

The organizers of the annual Burning Amplifier Festival are busy gearing up for this year's event to be held October 19 at the Presidio Yacht Club in San Francisco, Calif. The event promises an international group of audio DIYers, great sounds, great equipment, plus plenty of new ideas you can bring to your next project. Go to www.burningamp.com for more information and to register.

CAUTION
BURNING AMP

MIGHTY MIDGET

In the interesting Mighty Midget article by Peter Millett (May '08, p. 17), the idea of using a single tube to build an entire power amplifier has taken a new direction through the ingenius use of a TV double pentode tube. The way the schematic was developed, the clear and detailed explanations in the text as well as the practical implementation of the amp makes it really attractive and desirable to build. I suppose it has a special attraction to young or even not so young converts to the hobby, and for obvious reasons: simple and easy to build, low cost, and attractive. A huge crowd of entry-level young audio tube hobbyists is hanging around audio discussions groups on the web, and to these people, Pete's Mighty Midget must be music to the soul. Even if very few have direct access to *audioXpress*, there is always somebody in a discussion group to link a thread to an interesting article in the magazine, which has happened already with the Mighty Midget.

This kind of situation begs a question: Why not make the construction of this amplifier a little more accessible to this kind of crowd? A few examples: For some young lad without pertinent experience, the Japanese output transformers used in the amp may represent a nonattainable target; so why not mention a replacement, say, one of Hammond's 125 SE series available from AES even as the secondary reflected impedance is not exactly 7K.

Setting up the driver stage is a tricky task, as Pete mentions. Couldn't it be made less tricky? What about connecting the feedback path straight to the grid of the output tube through a small capacitor or to RC series combination? This way you may be avoiding even the unusual values of R2 and R4 and save some aggravation in the process.

I am well aware of the difference in quality between the two versions, at least in the measurement process, but does it make such a difference in the listening process? Many European radios built in the 50s and 60s used one or the other of the two versions in the AF output stage. You could find a few more similar samples in the article, but this is not the purpose of this letter. To be clear, I am not pleading here for a change in design; in my opinion this project is excellent the way it was conceived and it is understandable that any change or part substitution may alter the end product.

I suggest the author offer alternatives to make the project more attractive and feasible to entry-level hobbyists, even if it is going to sound a bit different from the intended version. In my view this kind of approach would apply to any practical project of this kind. Perhaps a part list including sourcing of the main parts would have also been beneficial.

Monny Nisel
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Pete Millett responds:

Thanks to Mr. Nisel for his kind words. As is true of most of the projects I publish, the Mighty Midget amp was targeted at those builders who are interested in DIY audio, but aren't willing or able to spend large amounts of money and effort on a project. I think this amplifier would be an excellent first project.

I'm afraid that Mr. Nisel must have missed reading part of the article. A sidebar, "Some Suggestions for Builders," included some practical suggestions, including some suggested transformers that are readily available, unlike some of the parts that I used.

As far as setting up the output stage and setting the values of the resistors in the feedback path, the "tricky" part is really just in the initial design of the amp; once the appropriate values have been determined, you should be able to reproduce it without any experimentation. I did use 1% resistor values, but I don't think there's much difficulty these days in finding 1% resistors, so I don't really see that as much of a problem. I also tested the circuit with a number of different tubes to make sure the performance was not too tied to individual tube characteristics, which vary quite widely.

That said, I have to admit that a 61.9k, 1W resistor, in particular, is a bit of an odd part (even though it is a "standard" value), which I didn't realize when I published the circuit. I probably should have mentioned that a 62k, 5% part would work just fine, and is certainly easier to find. Similarly, 357k could be replaced with a 360k, 5% resistor.

I have published full parts lists with vendor information for projects in the past. I've found doing so can be a bit of a two-edged sword. The problem is that if a particular part specified is out of stock, or discontinued, it really can cause more trouble for an inexperienced builder than if only a "generic" part is specified. For example, for the 0.22µF 400V coupling capacitor: if you search for "0.22µF 400V" part at one of the catalog distributors, you will probably come up with many options, any of which would work fine. But if I were to specify a particular vendor catalog part number, and it happens to be out of stock, what does an inexperienced builder do?

GOING GREEN

The CES 2008 went green for the first time by promoting audio products using recyclable or green materials—an indication that the industry is starting to pay attention to the undeniable threat of global warming. It is the responsibility of the audio industry to green up, as other industries already are (e.g., the Green Computing initiative by the computer industry). Designing and manufacturing audio products in a way that minimizes the impact on the environment is a very important step.

Green Audio? The next, much more important step, is how we audiophiles make our contribution. Let's acknowledge that the effort to produce eco-friendly products is a drop in the ocean compared to the environmental impact of *using* these products over time. Many of us enjoy our audio systems, for several hours every day, every week, every month, and that translates into many KWH (KiloWattHours) of electricity needed to operate our audio systems and that's a way bigger threat to the environment than production of audio equipment itself. . .

Mike Zivkovic
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This letter continues at www.audioXpress.com.

SOLDERING

I was pleased to see Ed Simon's article ("Soldering: A Tutorial," March '08, p. 42): Soldering is a very important topic that is given far too little attention today. I figured I'd add a few comments about irons, and a few more tips about soldering. I was an assembler and an inspector on mil-spec electrical components many years ago, so I have had a bit of practice—and some experience—with what can go wrong if you don't do it right.

First, I should say that anyone who expects to do any serious soldering should

CONTRIBUTORS

After receiving his technical training in the Air Force in the mid 1960s, **Rick Spencer** ("A Beginner 6BQ5 SE Amp," p. 6) has worked in the air conditioning, heating, and refrigeration systems field ever since. He has built over 30 audio components in the last 41 years and is currently working on at least two amplifiers that will be low in cost but high in sound quality (and they are geared toward the novice who is looking to get started in this wonderful hobby).

A specialist in a practical approach to electronics, **John Laughlin** ("An Automated Level Control," p. 18) has spent 25 years in the electronic industry, and 25 years as a college instructor. He has a first-class radiotelephone license, and is a member of the Brazos Valley Amateur Radio Club. He has written for several publications, including *Popular Electronics, Ham Radio Magazine*, and *QEX*. Besides electronics design, he enjoys making custom billiard cues, sailing, and he has walked across the Grand Canyon three times.

The audio hobby for **Alex Arion** ("A Greek Triad," p. 24) began in the magic '60s in Communist Romania in his grandmother's attic, where, together with two friends, he installed a small "illegal" laboratory. Their first attempt was a beginner's guitar amplifier and a professional guitar pickup. The tubes used included ECC40s and EF40s. The rectifier was a 5µ4, Russian equivalent. They mounted all in a hat box that they found in the attic. After finishing his studies and many years of research, Alex moved to Greece, where he continues to build different tube-based and solid-state amps.

Joseph D'Appolito ("Testing Loudspeakers: Which Measurements Matter, Part 2," p. 28), regular contributor and author of many papers on loudspeaker system design, holds four degrees in electrical and systems engineering, including a Ph.D. Previously, he developed acoustic propagation models and advanced sonar signal processing techniques at an analytical services company. He now runs his own consulting firm specializing in audio, acoustics, and loudspeaker system design. A long-time audio enthusiast, he now designs loudspeaker systems for several small companies in the US and Europe.

Ed Simon ("An Isolation Transformer, Pt. 3") received his B.S.E.E. at Carnegie-Mellon University. He has installed over 500 sound systems at venues including Jacob's Field, Cleveland, Ohio; MCI Center, Washington D.C.; Museum of Modern Art Restaurants, New York; The Coliseum, Nashville, Tenn.; The Forum, Los Angeles; Fisher Cats Stadium, Manchester, N.H.

John Shand ("Jazz Track," p. 54) is a CD reviewer for *Australian HiFi and Home Theatre Technology.*

invest in a real temperature-regulated iron—one that uses an actual sensor and feedback mechanism. Not only will it heat faster and make the job easier, but it will actually make you much *less* likely to damage expensive components and PCBs, and improve your results dramatically. Also, being less fussy, it just makes soldering much more fun. . . .

Keith Levkoff
kLevkoff@panix.com

This letter continues with Keith Levkoff's insights and soldering tips for both beginners and experts on our website, www.audioXpress.com.

CROSSOVER APPROACHES

I'd like to thank Mr. Koonce for documenting his research and presenting the results of his efforts ("Passive Crossover Linear Phase Speakers," June/July '08). I really liked the special plot format—he developed a satisfyingly high information density, as Edward Tufte would say. The problems with designing a linear phase (transient perfect) multi-speaker system are clearly seen, and I must admit to having gone to the dark side and adopted DSP to implement my crossovers in the digital domain for exactly these reasons.

However, the problems with lobing in multi-way speakers remain, and the discussion of coaxial speakers was very interesting and timely for me. From the plots for the coaxial speaker, Fig. 7 in particular, the magnitude and phase responses are very similar to those of the modified two-way in Fig. 5. Could Mr. Koonce confirm whether he used an actual coaxial driver for the real driver plots in Fig. 7 or whether he simulated a coaxial driver using the responses of the woofer and tweeter shown in Fig. 3 by simply setting the vertical offset to 0?

The reason I ask is that, as Mr. Koonce states, you need to ensure that the acoustic origins are in the same point in 3D space, otherwise there will always be some angle where there is a phase difference between the two drivers at the crossover point and lobing will occur. In fact, in a coaxial speaker, having one driver in front of the other acoustically is worse than a vertical or horizontal offset because it will cause lobing in both vertical and horizontal responses. I have been pondering the experiment to test whether coaxial drivers can make their acoustic origins coincident. I'm not sure that they can, and I also don't think that adding delay will compensate either, because different angles require different delays.

Another factor I have been considering is how frequency plays into this. It would seem that high frequencies, with their shorter wavelengths, would be more susceptible to a particular offset. To illustrate; a 30mm offset represents a 94° phase shift at 3kHz, but only 9° at 300Hz. This seems to imply that the lobing should be reduced with a lower crossover frequency.

A couple of final questions: It wasn't clear in the text, but I'm assuming that you used the acoustic phase responses of the speaker drivers as well as the zobel networks in the models. I haven't simulated any speaker-crossover systems, but I would expect that the zobel network would affect the speaker acoustic phase response as well.

Did you use an electronic circuit simulator to model the crossovers-speaker system or did you just crunch the math in a spreadsheet? Before I set out on my own math crusade, I'd be interested in hearing your opinions on the best environment in which to work.

Thanks again for a great article.

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G.R. Koonce responds:

I want to thank Iain McNeill for the kind comments on my article concerning linear-phase speakers. All "real" two-driver-response systems used the two responses shown in Fig. 3 and all "real" three-driver-response systems used the three responses shown in Fig. 10. Thus, the coaxial driver results shown in Fig. 7 are for the real driver response plots of Fig. 3 with the vertical offset set to zero. I apologize for not making these points clear in the article.

You can build coaxial drivers with an optimum alignment of their acoustic origins. This is accomplished by building the tweeter behind, or inside, the woofer motor and firing right through the woofer center.

However, just because the tweeter motor is located behind the woofer's cone does not ensure proper acoustic origin alignment. I have measured some that were terrible in this regard. I also worry about Doppler distortion with coaxial drivers that use the woofer cone as a horn for the tweeter. The car coaxial drivers with the tweeter mounted in front of the woofer cone generally have not tested well for acoustic origin alignment.

Adding true time delay to the tweeter is effectively the same as moving the tweeter backward horizontally. With a coaxial driver the correct time delay to align the two acoustic origins will be correct for all vertical and horizontal angles. The ability to use time delay is a major advantage of using a digital crossover implementation. It allows building a phase-linear system without having to "step" the front panel. However, even with time delay, only the coaxial driver will avoid lobing problems. A multi-driver system will still have linear phase over only a limited vertical angle (assuming a vertical array of drivers). As developed in the article, the use of a symmetrical (D'Appolito) array greatly improves performance, but not to the extent possible with a properly aligned coaxial driver.

It is true that the effect of acoustic origin offset becomes worse with increasing frequency; i.e., with shorter wavelengths. Thus generally with a three-way system the lower (woofer to midrange) crossover causes fewer problems than the upper (midrange to tweeter) crossover. With a two-way system, whether things become better with a lower crossover frequency depends on the drivers involved and should not be taken as a general rule-of-thumb. When going for a linear- or minimum-phase system, the driver-response overlap requirements may not allow much liberty in crossover frequency selection.

My approach to modeling is as follows. A modeling program I wrote takes in the driver's on-axis acoustic response (magnitude and phase) and input impedance (magnitude and phase). The program is also fed the physical placement of the drivers including the acoustic origin positions. The crossover is added as components, and the program computes the response of each crossover section loaded by its driver and zobel, if used. Thus, the zobel will affect the voltage (magnitude and phase) feeding a driver.

The directivity of the driver (change in its response with angle off its own axis) is modeled by the program. If desired, the effect of diffraction spreading loss is also modeled on each driver's acoustic response. Finally, the acoustic response for each driver gets a phase shift correction for the path to where the system response is summed. This is all done at each frequency that is to be plotted. Thus, the approach is a lot of number crunching done in a program written just for that purpose.

Many systems were breadboarded in my test baffle to verify that the modeling does a good job of predicting the response. This approach is surely not the only way to solve the passive crossover design problem, but is one that works for me because it fits what I can measure about the drivers I intend to use. You could use a similar modeling approach with an active crossover and multiple amplifiers with the simplification that the driver's input impedance would not need to be considered. Voltage-sensitivity differences between drivers would also be handled by amplifier gain correction rather than by padding networks in the crossover. aX

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Ad Index

Jazz Track

By John Shand

John Abercrombie|The Third Quartet|ECM 1993

For seven years John Abercrombie has plowed this quartet's field of possibilities, and the sonic similarity between his guitar and Mark Feldman's unamplified violin actually increases the air in the music. While they soar to their instruments' upper reaches and circle each other like two blue kites, bassist Marc Johnson seems gravitationally drawn to the bass's lower register, and all the space in between seems to further the illusion of an updraft. Drummer Joey Baron is too astute to clutter this. He keeps his sounds as concise as his ideas, and mostly espouses them softly. Given such clearly defined roles, and despite the unfettered improvising, the end effect is not unlike a string quartet, dancing as one across Abercrombie's wistful melodies, plus one each by Ornette Coleman and Bill Evans. Superb.

Oynsemble Melbourne|Ascension|Birdland BL 012

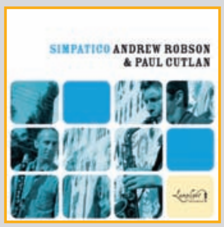
If you've ever stood in a tropical storm and relished the deluge rather than trying to escape it, you already know something of the pleasure contained in the torrential power of Ascension. The stimulus was the two 1965 molten pieces of music recorded under this name by John Coltrane. Melbourne drummer Ted Vining has taken Coltrane's simple theme and collective fervor and not just run with it, but bolted. The original's 11 players have become a gargantuan 19, perhaps over-complicating the recording, because the mix is too subdued. This is at odds with the actual music, which is sometimes an exhilarating experience, as surging solos and duets are punctuated with collective squalls. Among those to have ascended the storm clouds to this particular heaven are saxophonists Andy Sugg and Adam Simmons, and trombonist Adrian Sheriff.

Adrian Cunningham Quartet|In Motion|Newmarket NEW3229.2

Many musicians, like actors, have a wretched time being pigeonholed, but Adrian Cunningham has a whole wardrobe of multiple musical hats. A rarity, he is equally convincing across the whole jazz spectrum, and his own band—with pianist Bill Risby, bassist Dave Pudneyn, and drummer Gordon Rytmeister—only marginally narrows the scope. Take the first three tracks of this live CD. *Keeping Fit with Ken* has Cunningham playing tenor saxophone on a high-energy 24-bar blues in the 60s Blue Note mold. The second, *Ubirr*, places his clarinet in a context of such lyricism it could spring from the ECM catalog. The third, *Gratitude*, represents the West Coast big ballad school—and is therefore my least favorite piece on a very entertaining album that is accompanied by a bonus live DVD.

Wynton Marsalis|From The Station To The Penitentiary|Blue Note 094637367520

Most artists either over- or under-estimate their own abilities. It's the ones capable of accurate self-criticism who usually produce the best work. In creating a song-cycle critique of modern America and the plight of its black population, Wynton Marsalis's motives were sound, but he is not the lyricist and barely the composer to realize his vision. The big plus is that his writing—sometimes reminiscent of Charles Mingus—leaves much improvisational space, and the playing is as exceptional as the recording quality. Marsalis himself blows some unusually raw trumpet, and saxophonist Walter Blanding sends emotional tornadoes through several pieces, scooping up the other players, including buoyant pianist Dan Nimmer. Young singer Jennifer Sanon, however, is a little green to be saddled with the challenge of making the words convincing.



Andrew Robson & Paul Cutlan|Simpatico|Lamplight LLR00106

Just the thought of horns without accompaniment is enough to drive some people to distraction, or at least out of the room. They should risk lingering a little. Andrew Robson and Paul Cutlan have been playing together for a dozen years, and their *simpatico*, always conspicuous, now flourishes on 11 improvised duets for Robson's alto saxophone and Cutlan's Eb and bass clarinets and tenor saxophone. There's nothing esoteric or "difficult" about this beautiful music. Texturally it is as abundant as a rain forest, with Cutlan's bass clarinet being a garden of surprises all by itself, and the fused musicality of the two players ensures the improvisations are never groping for ideas. Each piece is distinctive in mood, from the braying eastern carnival of *The Mighty Khan* to the boppish swoops of *Running Into Time*.

Tord Gustavsen Trio|Being There|ECM 2017

Tord Gustavsen's third album luxuriates in more of the spacious, softly contoured, unashamedly romantic jazz, the pianist's compositions performed with bassist Harald Johnsen and drummer Jarle Vespestad. Being There is not just a carbon copy of its predecessors, however. Where they were kept on a tight dynamic leash, this one brings a nudge more of the drama implicit in the trio's concerts to the surface, moving the music further away from an "ambient" tag. As before, the pieces are kept short and the melodies are often exquisitely sensual, while the harmonic modulations can sometimes add the vaguest trace of soft-porn corniness. Gustavsen has beautiful touch and a facility far beyond the needs of his own compositions. Vespestad again exhibits almost freakish control, his drums and cymbals wearing a halo of reverberation.