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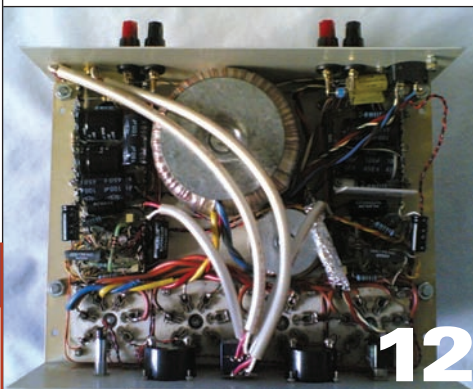
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Beginning with this issue, digital versions of **aX** will be available to our readers. To subscribe, send your e-mail address to Sharon at custserv@audioXpress.com. Besides the material covered in the printed version, digital versions will include additional articles. The February issue includes Jan Didden's report from New York on the AES show, and Martin Daley puts together a low-cost, high-end music server.

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Bridge to a Voltage Doubler

With a simple connection, you can convert a full-wave bridge rectifier I-C (such as the Radio Shack 276-1185) to a voltage doubler. The advantage here is the high current rating. For example, the 276-1185 with its 25A rating can provide 12.5A (in this configuration, the IC's rating is halved).

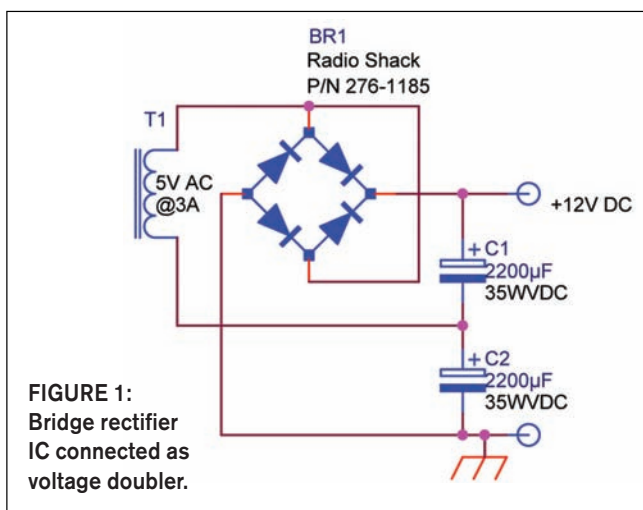
Figure 1 shows how to rewire a bridge I-C as a voltage doubler. It is a circuit I used to put an extra 5V AC, 3A rectifier filament winding to use as a source for heating 12.6V A-F and total of 0.9A, which is the same as

two 5687s, or five 5751s.

In vintage hi-fi amps, the Radio Shack 276-1173 can be used as a re-

placement for the common axial diodes in voltage doubler power supplies. *aX*

Neal A. Haight
Castro Valley, Calif.



Parts List for Fig. 1

T1 5V, 3A
Hammond P-T167 M5, available from Antique Electronic Supply (or can be extra 5V AC, 3A winding on power transformer)

C1, C2
Electrolytic axial capacitors
2200µF/35 WVDC, Radio Shack part no. 272-1020

BR1
bridge rectifier IC, 50V 25A, Radio Shack part no. 276-1185

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Laminated MDF Three-Way Speaker System

This layered construction results in a professional-looking box with high-end sound.



One of my favorite acoustical engineers is Harry F. Olson, whose work on diffraction¹ influenced this three-way project. I wanted to construct a good-looking, practical, living room loudspeaker, with the least possible sound compromises, using some of the most popular units of the 80s that were made in England by KEF.

DRIVER SECTION

The B139 is an oval-shaped woofer with an effective piston area equivalent to that of a 12" conventional one, but with

a better dispersion pattern. It has a die-cast aluminum basket and develops all its quality and power into a relatively narrow bandwidth. Mainly, it was used in TL (transmission line) configurations.

The B110 is a midwoofer able to stand alone at low frequencies, also used in many famous designs either as a woofer or a midrange. The crossover point for this multi-driver speaker is around 150Hz, benefiting the middle spectrum. It is also interesting that their voice coils lie on the same plane.

Finally, T33 is a classic 1" textile dome tweeter with a natural flat sound, but it is considered by many not to be in the same class as the B139 and B110. I used acoustic suspension (AS) for both drivers.

OVERVIEW

The advantages of AS are: very good attack, flatness, low-end extension, and less diffraction. The disadvantage is the low SPL and is compensated by the three-way/12" woofer configuration. Another issue is the relatively low efficiency of the KEFs, especially the B139. That is compensated by the active crossover/multiple amps system.

I built two box sections per driver with laminated flanges of 22mm MDF (medium density fiberboard): one for the woofer (low-frequency box, LFB) and one for the midrange (mid-frequency box, MFB). The modified, spherical T33 tweeter (**Photo 1**) is mounted on the top of the MFB. I installed a sheet of 1mm lead² around the internal vertical surfaces, to further damp wall resonances and attenuate the mid frequencies escaping out.

Because upper and lower panels are parallel, I perforated them in such a way as to produce air columns resonating in a different frequency, in an effort to reduce standing waves. At the same time,

a kind of honeycomb is produced, which responds effectively to the resonances of those panels.

The six sidewalls are, of course, not parallel. Inside the unit, the extensive chamfering of the drivers' mounting holes and the unstuffy diverge part provide an easy path to the converge part. Back there most of the energy of the sound waves is lost by hitting the side walls traveling through effective damping, before they manage to return to the speaker cone.

Lower panels are removable for better access and to allow for an adjustable slot for a future aperiodic configuration experiment.

After the break-in procedure, I evaluated all driver parameters with a frequency generator and oscilloscope. Referring to the *Loudspeaker Design Cookbook (LDC)*³, I determined the Q target values around .8 for the LF box and around .5 for the MF; the volumes should be 70 ltr and 7 ltr, respectively.

DESIGN CONSIDERATIONS

You need to consider the following, with



PHOTO 1: The finished three-way speaker.

regard to the shape of the MFB:

1. The modification on the T33 tweeter regarding the distance between its point of radiation and that of the B110⁴.
2. The distance between the MF and LF units, accounting for the height of the spikes.
3. The dimensions and the shape of the flange, which, in turn, depend on:

- a. Wall thickness, which is thicker at the speaker-mounting baffle, at the corners, and at the place on the larger surface where a mass or brace is produced.
- b. External shape to be a sort of truncated double pyramid¹.
- c. Internal cutout to diverge just aft of the speaker unit and then converge.

- d. Inside volume to be 7 ltr, accounting for the perforated upper and lower panels, the speaker unit, and the stuffing.
- e. The MFB should come out of the LFB, in order to save material.

The same basic principles apply to the LFB. In addition:

1. Its height should place the MF/HF

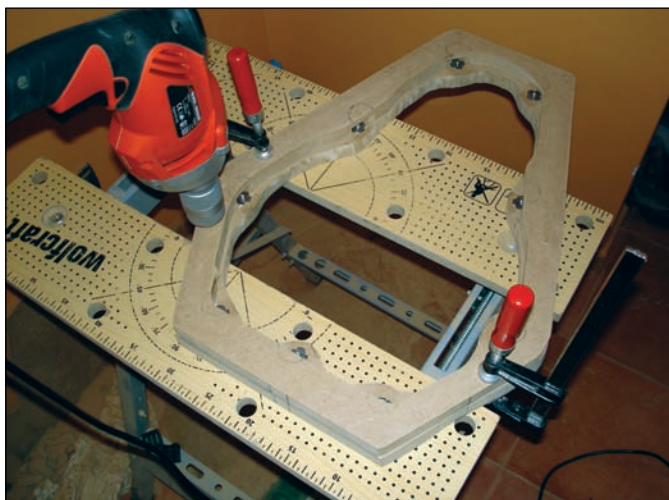


PHOTO 2: Starting with the base of the unit.



PHOTO 3: Affixing the layers.

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PHOTO 4: The woofer box with sheet of lead installed.

- acoustic center at the level of the sitting listener's ear.
2. Front baffle width should be about the woofer's size, which measures 21cm.
3. The depth should be around 50cm, which is a nice dimension, because even in close to the wall there is about 55cm to 65cm distance from the LF radiating units⁵.
4. In order to save material and cutting effort, these LFB flanges should fit next to the other in three rows, when sketched on the MDF board.

CONSTRUCTION

You need a breathing mask and eye protection when cutting or sanding MDF, which contains adhesives that may cause eye and lung irritation. I used a disc saw for the externals and a table-saw for the internals. The LFBs include

62 layers, while 18 layers make up the MFBs.

I numbered pieces bottom to top, from #1 to #31 for the LFB and from #1 to #11 for the MFB. I mainly used 22mm thick MDF, but 10mm MDF for levels #1 and #11 of the MFB.

I used a heavy-duty table drill equipped with a quality 6cm diameter circular cutter to perforate levels #2 and #30 of the LFB and levels #2 and #10 of the MFB, except with smaller holes. Flange #3 at both boxes is wider inside, to carry the T-nuts for the bolts holding the lower panels. Also trim #4 to allow for T-nuts replacement if needed and for the extended length of the bolts threads.

As soon as the first flanges were at hand, I began the assembly process. Working with 25cm clamps, I prepared three groups of nine layers for each



PHOTO 5: Stuffing the woofer.



PHOTO 6: Positioning the woofer driver.

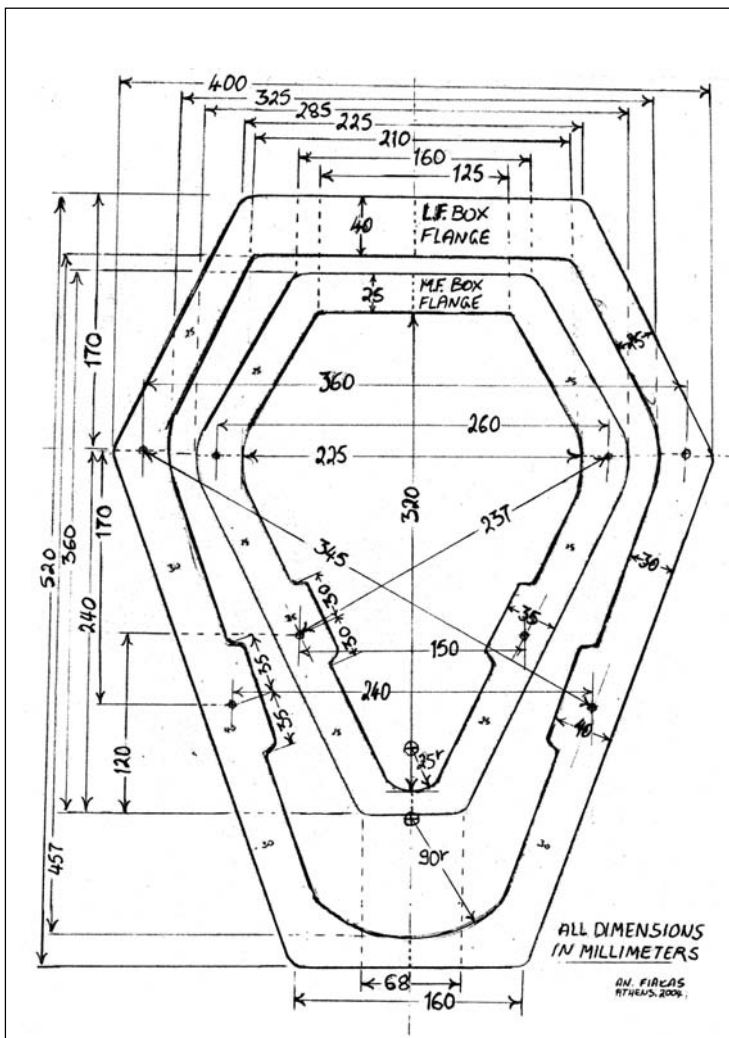


FIGURE 1: Woofer and midrange flange dimensions.

LFB. I used longer 80cm clamps, until the entire body was completed.

I aligned the pieces using a ruler, a right angle, a level, and even a plum bob, depending on the situation. They were held in place with a couple of clamps, and I drilled two opposite 3mm holes. I applied glue to both surfaces, and aligned them by inserting two 3mm drill bits into the holes. I then applied clamps every 10cm, retightened them after 20 minutes, removed the bits, and let the pieces set for 24 hours. The PVA wood glue I used is water resistant and white, which then turns transparent while drying.

I flattened the front baffle and, using a router, produced the recess for the drivers to be flush mounted. Then I marked the speakers' holes, cut, and chamfered the inner edges. Next I installed the already cut-to-fit sheet of lead, using 10mm wood screws and plenty of silicone adhesive.

ASSEMBLY

At this point I glued the top panels and attached the lower panels with the bolts. I rounded the corners and with a flat wood file and sandpaper gradually worked the outside to a smooth finish.

Then I applied four layers of matte and two layers of gloss transparent lacquer with a brush, in an environment of low humidity and elevated temperature, and smoothed with super-fine sandpaper between applications. If a repair action was required, I mixed leftover MDF particles with matte lacquer or a primer to form a kind of cement.

I used two materials for stuffing—lamb's wool and foam damping board, both purchased from Visaton (www.Visaton.com). Flat foam blocks in the LFB keep the damping boards 2cm from the walls and fill the holes of the upper and lower panels. I used these blocks as well in the MFB, except the distance is smaller. Wool fills in the rest of the space, except at the area around the units. I then loosely filled the LFB and MFB with foam damping boards.

After removing the four bolts, carefully separate the tweeter from the factory flange. You could have a machinist construct a new aluminum alloy mount for a better look, or you could make your own.

I chamfered the opening for the tweeter dome by hand and finished it with metal polish. Then I used a dremel tool to open and rout paths for the wires. There are three holes for the 3mm bolts that fix the new unit on the top of the MFB; two of them located under the magnet are threaded.

The sphere is made of polystyrene foam, which I purchased from a decorations shop and trimmed it to fit. The Linkwitz active crossovers have 24dB slopes and cross at 170Hz and 2.6kHz. The McLelland CX-154 adjustable active crossover is also in use, especially during the tests.

Three power amplifiers per channel assembled using ZAPpulse class D modules operate a total of 650W RMS for a safe power backup. Class D amplifiers perform better with heavier transformers than bigger caps, so I chose the following: woofer amp transformer—650VA Giatras (high-quality local manufacturer); caps— $2 \times 15000\mu\text{F}$ Sic-Safco Felsic 85; midrange amp transformer—500VA Giatras; caps— $2 \times 15000\mu\text{F}$ Sic-Safco Felsic 85; tweeter amp transformer—200VA Giatras; caps— $4 \times 4700\mu\text{F}$ Philips.

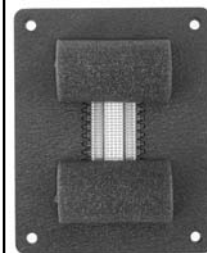
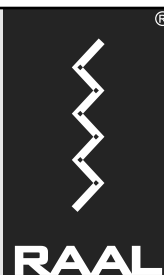


PHOTO 7: Spikes on the base of the unit.



PHOTO 8: The completed woofer section.

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- Frequency response: 1.6kHz to 100kHz
- Sensitivity: 95dB /1m / 2.83V

Both tweeters also available with Amorphous cores for about \$100 more each.



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PERFORMANCE

Building this kind of speaker takes a lot of time and materials and it is not possible to make alterations. So I breathed a sigh of relief when, during the primary measurements—made before the first listening tests—the Nyquist plots appeared as almost circles for the empty boxes. Further measurements of the impedance and the complex impedance

carried out (accomplished) with minor discrepancies. With a moderate stuffing, the Q for the LFB was .82 and .47 for the MFB.

The finished unit is a high-end loudspeaker with a serious capability of sound reproduction, especially at the lows. Low frequencies suggest an open baffle configuration, but with better decay.

Very good transients⁶ are everywhere; for example, a Pink Floyd fan was surprised at the beginning of “Breathe,” from *The Dark Side Of The Moon* album. Classical music orchestras are a pleasure to listen to, as multiple instruments hold their positions and their identities. However, sometimes the mids appear not so airy. I may need some time for further trimming.

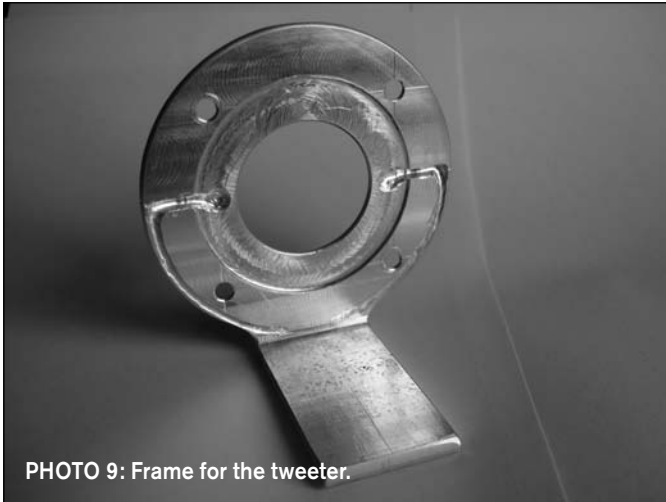


PHOTO 9: Frame for the tweeter.

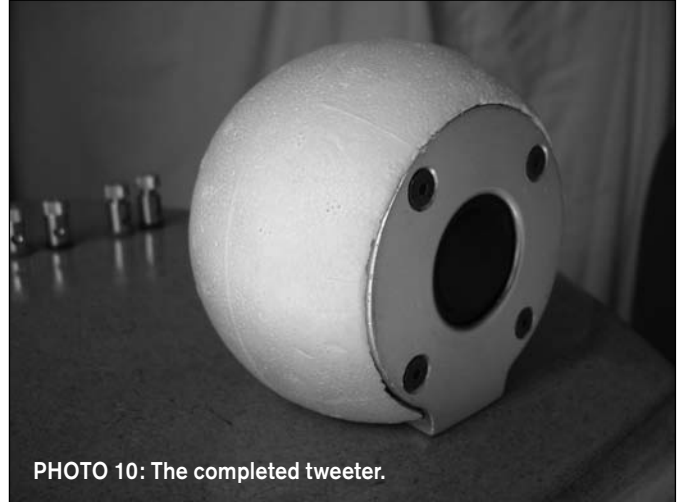


PHOTO 10: The completed tweeter.

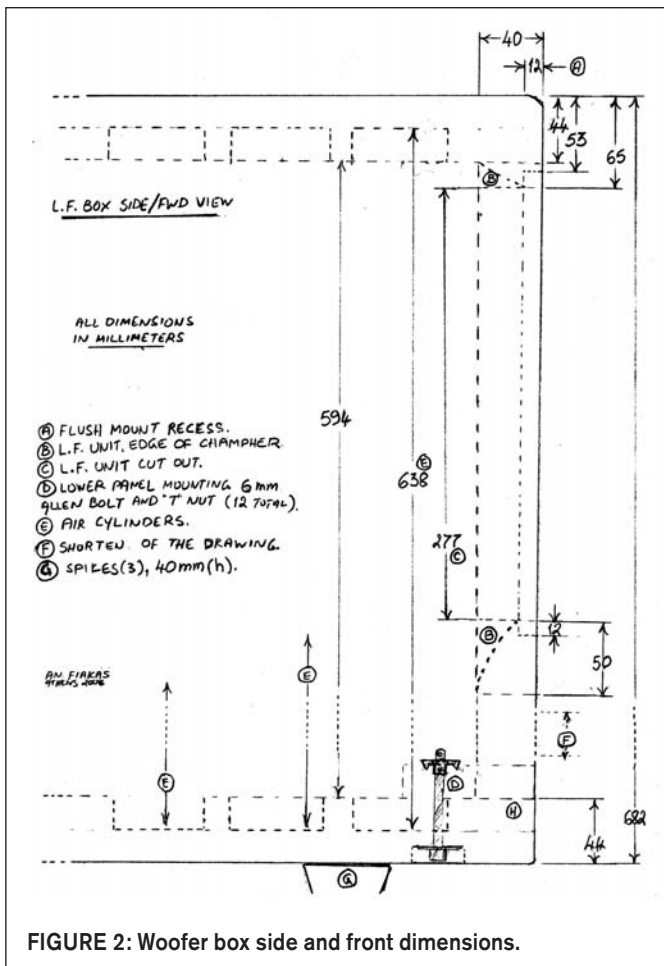


FIGURE 2: Woofer box side and front dimensions.

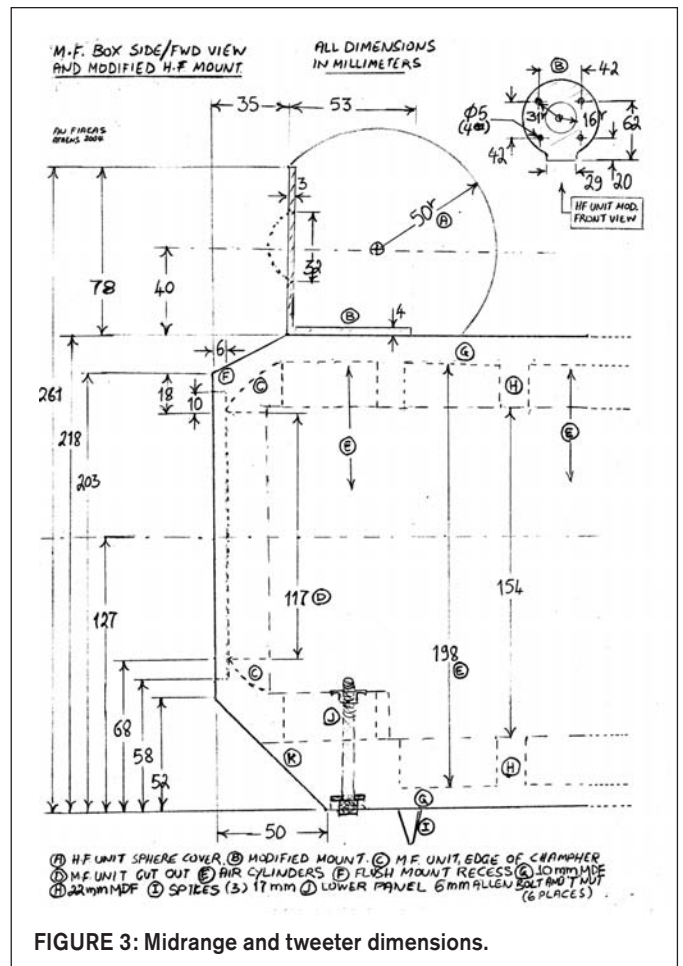


FIGURE 3: Midrange and tweeter dimensions.

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1. *Acoustical Engineering* by Harry F. Olson*.
2. *The Master Handbook of Acoustics* by F. Alton Everest*.
3. *The Loudspeaker Design Cookbook* by Vance

Dickason*.

4. *Great Sound Stereo Speaker Manual* by David B. Weems. First Edition, 1990. TAB Books.
5. *Loudspeaker Anthologies* from the Audio Engineering Society*.



PHOTO 11: First three layers of midrange section. Note perforations in board # 2.



PHOTO 12: Finished speaker and amp.

6. *Loudspeakers* by G. A. Briggs, assisted by R. E. Cook. Wharfedale Wireless Limited. 1958 (336 pages).

* Available from Old Colony Sound Lab at www.audioXpress.com.

PHYSICAL CHARACTERISTICS

Height: 1035mm (40.75")

Width: 395mm (15.55")

Depth: 515mm (20.27")

Weight: 46.7kgs (102.74 lbs)

Weight of the satellite alone is 8kgs (17.6 lbs).

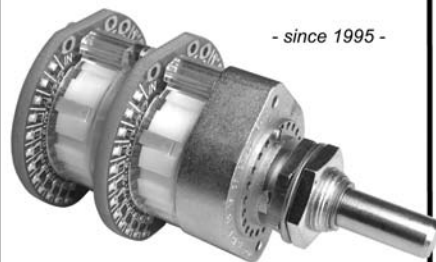
Note: The dimensions may be slightly smaller due to the sanding.

ACKNOWLEDGMENTS

I would like to thank Dionisis Sigiridis for making the state-of-the-art tweeter mount and George Papageorgiou for making the active crossover and for his assistance through his laboratory in testing and developing the speakers and amplifiers. *aX*

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Presenting a new output-transformerless circuit with direct coupling to the loudspeaker.

If you have spent \$500 or more on 5m or so of exotic loudspeaker cable, have you ever wondered about the 500m of standard copper wire in the output transformers of your tube amplifier? Audio output transformers are large, expensive components that require complicated winding arrangements in order to work properly at high frequencies. They are the prime culprits for the soft bass sound associated with tube amplifiers.

The main causes of this are iron core saturation distortion and the winding inductance which bypasses the loudspeaker at low frequencies. Also, the winding resistance typically wastes 10% of the output power. Hence, a lot of iron and copper are required in order to minimize these problems.

An alternative is the output transformer-less (OTL) tube amplifier. However, this concept is not easy to realize in practice, otherwise there would be more of these around.

AMP OPTIONS

My OTL design offers several solutions. First, in order to protect the loudspeakers in the event of a fault, it needed to be naturally current limiting without using auxiliary protection circuits. Another problem was how to realize a symmetrical output stage when tubes do not come as complementary NPN and PNP pairs as with transistors. One option was to adopt the “circlotron” circuit¹, which was invented by Cecil Hall in 1951, but that precluded the use of natural current limiting and would have greatly complicated the power supply configuration.

Instead, I designed a non-complementary totem-pole output stage using a novel combination of local feedback and current drive in order to achieve good symmetry and cancellation of even harmonics, as confirmed in subsequent measurements. This configuration has

more in common with the Futterman circuit², except that a long-tailed pair of pentodes is used for the driver stage instead of the concertina phase splitter. The pentodes provide the current drive as well as greater voltage swing than triodes.

A general aim of the design was to have as simple a circuit as possible with a minimal number of components in the signal path as well as push-pull operation throughout (Fig. 1). Push-pull amplification not only cancels even harmonic distortion, but also provides good rejection of power supply ripple. In a long-tailed pair, the supply current is virtually DC so that the power supply is effectively removed from the signal path.

Above all, I wanted a stable, reliable design that would not need constant readjusting. To this end, I incorporated

ample loop DC feedback, which—after initial adjustment—keeps the offset voltage within 20mV between tube replacements. Similarly, the DC bias needs hardly any adjustment over time.

I know that signal feedback is a controversial issue and there are those who maintain that the ultimate goal should be 0dB. However, zero feedback in this design would result in audible noise and an output impedance of 8Ω, which would severely affect the tonal balance of most loudspeakers. I have applied 26dB of feedback, which is a similar amount to most classic tube designs and sets the output impedance to 0.4Ω for well-controlled bass. However, the advantage of a DIY amplifier is that you can adjust the feedback to suit your own taste. The simplest way to reduce the feedback to 11dB is to omit the cou-

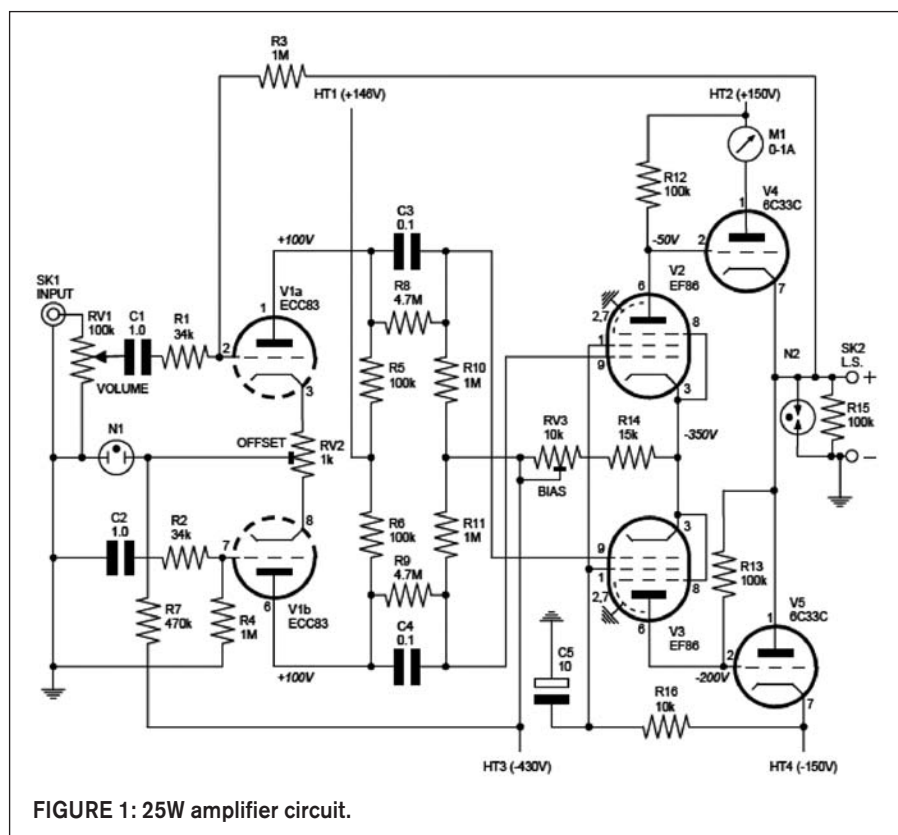


FIGURE 1: 25W amplifier circuit.

pling capacitors between the first and second stages.

Finally, in order to drive normal loudspeakers, I decided that I needed a power rating of at least 20W. The obvious choice of output tube was the Russian designed 6C33C triode, because a single pair can deliver 2.5A into an 8Ω load from a moderate 150V rail. This enables the amplifier to deliver 25W into an 8Ω load or 40W into a 16Ω load, such as my full-range reflex-port loaded Lowthers. If you can increase the load to between 40 and 100Ω, then you can easily obtain 50W of pure class A power.

I could only measure the distortion without feedback (by injecting the signal directly into the grid of the input tube) because the distortion with feedback was less than that of the signal generator. This gave 0.14% THD at 2W with an 8Ω load without feedback, or 0.007% with 26dB feedback. I am pleased to say that, during the eight years since I built this amplifier, only one fault has occurred, which was an internal short-circuit in one of the output tubes. Luckily, the HT fuse did its job and no further damage occurred.

I have designed and built many tube amps over the years from push-pull ultralinear to single-ended triode, using home-wound transformers. I have even experimented with solid-state, but I tend to be highly critical of my own

work and have never been fully satisfied with the results until now. This amp (**Photo 1**) just lets me enjoy the ambience and natural tonal color of a real performance. Unfortunately, it is somewhat unforgiving of recent highly compressed recordings, rather preferring early stereo classical and jazz vinyl LPs made with simple tube equipment.

CIRCUIT DETAILS

The signal from input phono socket SK1 is fed to the grid of V1a via the volume control RV1, C1, and R1. Shunt feedback is provided by resistors R1 and R3, which mix the output and input signals to set the overall maximum gain to a value of $R3/R1$, which is about 29. In other words, an input voltage of 500mV is required to produce 25W into an 8Ω load. When RV1 is set to maximum, the input impedance is about 26k due to RV1 in parallel with R1.

I included capacitor C1 to maximize DC feedback. When there is no offset, the grid of V1a is at the same potential as that of V1b, which is grounded via R4. However, small differences in grid to cathode voltages of each section, due to mismatch, can produce a voltage at the grid of V1a. This also appears across the loudspeaker as a DC offset because the 100% DC feedback path, via R3, keeps the input and output voltages equal. You can adjust trimmer RV2



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AP-016



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The AP-016 is available in three versions. The AP-016A crossover point is at 2Khz, the AP-016B is at 2.7KHz and the AP-016C is at 3.5KHz. All of them are 4th order Linkwitz-Riley crossover.

Specifications:

HF Power Output: 30Wrms
LF Power Output: 80Wrms
THD: 0.03%
S/N ratio @ rated W: 90db
Input sensitivity: 1V
Input impedance: 22Kohms
4th order X-over: 2KHz AP-016A
2.7KHz AP-016B
3.5KHz AP-016C

Weight: 2.6Kgs (5.7lbs)
Dimensions W x H x D:
W: 137mm (5.4")
H: 218mm (8.6")
D: 81mm (3.2")
Cut-Out W x H:
W: 108mm (4.25")
H: 190mm (7.5")
AC Voltage: 115V / 230V

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PHOTO 1: Front view.

to null the offset.

An alternative arrangement is to apply the input signal to the grid of V1b with series feedback applied to the grid of V1a. This has the advantage of allowing higher input impedance (1M, for example). However, it would also tend to unbalance the input stage slightly, unless you use an "ideal" solid-state current source in place of R7. Of course, you could achieve excellent noise immunity by using a preamplifier with a balanced output and then applying this to both grids of V1 via C1 and C2.

The input stage, which is formed by V1 and its associated components, acts as a long-tailed pair phase splitter. The high DC voltage across R7 provides a near constant-current that is split between the two halves of V1. This means that if the current through one half *increases* by a certain amount, the current through the other half must *decrease* by the same amount in order for the sum of the two currents to remain constant.

As these currents also flow through the anode load resistors R5 and R6, the output voltage developed across one of

them must also *increase* by the same amount that the voltage across the other *decreases*. An attractive feature of long-tailed pairs is that the current drawn from the supply is near constant DC. In other words, the supply is largely excluded from the signal path, and this reduces its effect on the sound quality.

The neon bulb N1 serves to limit the heater-cathode voltage on both halves of V1 to about 65V maximum during warmup. It is unlit during normal operation.

The balanced outputs from the input stage are coupled to the grids of V2 and V3 via C3 and C4. There is also partial DC coupling via R8 and R9. The driver stage, formed by V2 and V3 and their associated components, also acts as a long-tailed pair. The outputs of this stage are directly coupled to the grids of V4 and V5, which form the output stage. Trimmer RV3 allows the voltage developed on the grids of V4 and V5 to be adjusted in order to set the output stage bias current.

The choice of bias current involves a trade-off between tube life and distortion.

In theory, you can bias the output tubes up to 400mA maximum, at which point their anodes dissipate 60W. This gives the lowest distortion. However, you can achieve a much extended tube life with a lower bias current of, say, 200mA. This also reduces the considerable amount of heat produced by the amplifier!

I used pentodes in the driver stage because they can swing greater voltages than triodes and also because they make excellent current sources. The latter ensures symmetry within the output stage. Another benefit of the pentode is the virtual absence of Miller capacitance between the anode and control grid, due to the presence of a screen grid. This increases the bandwidth of the stage and eliminates the need for frequency compensation components in order to make the amplifier stable when feedback is applied. The only disadvantage is that they produce slightly more odd-order harmonic distortion than triodes.

However, the EF86 was designed for audio and, as such, is more linear than, for example, a variable-mu RF pentode. It was used very successfully in the driver stage of the famous Quad II amplifier. You can see the need for a current driver by analyzing the output stage.

V4 is a cathode follower. This means that there is 100% negative feedback between the cathode and the grid, which results in less than unity gain and reduced output impedance. V5 is an anode follower and, in order for it to have the same gain and output impedance as V4, it must have 100% negative feedback between the anode and grid. This is achieved by using a current driver, which, by definition, has very high source impedance that does not attenuate the feedback, which is routed via R13. Although the DC voltages on the anodes of V2 and V3 are not the same, this does not really matter much because, for a given grid voltage, pentodes tend to produce an anode current that is largely independent of the anode voltage. That, after all, is the definition of a current source.

R15 ensures that the grid of V1a is always grounded during warmup, in case no loudspeaker is connected. The gas discharge tube, N2, ensures that the output voltage is kept within safe limits.

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its under all conditions. If the output exceeds 90V, it strikes and then clamps the output to a low safe voltage until the output voltage has dropped below this (Fig. 2).

POWER SUPPLY

Although the power supply is fairly conventional (Fig. 3) and therefore in little need of description, there are a few points worthy of note: In the event of a fault forcing the output stage latch either up or down, R29 provides a means of limiting the current through the output stage and loudspeaker. If its value were too small, either an output tube or the loudspeaker or both could be damaged. If its value were too high, a small offset voltage across the loudspeaker could cause a significant imbalance in the supply voltages HT2 and HT4.

Fuses F1 and F2 are provided in the unlikely event that both driver tubes, V2 and V3, fail (or not being plugged in), thus causing an excessive current to flow through both output tubes V4 and V5. In theory, only one fuse is needed, but two are included here in order that any nonlinearity they produce is symmetrical. When connected directly in series with a loudspeaker, fuses are a known source of thermal nonlinearity. At low frequencies, their resistances vary with current to produce harmonic distortion. At higher frequencies, the thermal time constant is such that they produce dynamic compression. However, in this circuit, it is likely that the high internal impedances of V4 and V5 would swamp the small nonlinear resistances of the fuses. Possible improvements to this design could include the use of a DC supply for the heaters of V1 and a delay timer circuit to ensure that HT2 and HT4 are only applied when all the tubes have warmed up.

PRACTICAL CONSIDERATIONS

A list of components needed is shown in Table 1. The choice of smoothing capacitors C8-C11 is important because these definitely *are* in the signal path between the output tubes and loudspeaker and therefore should be of good quality. They should be free of internal vibration, which means that they should not “sing” when the amplifier is driving a dummy load. Unfortunately, since I

fitted the excellent Elna “cerafine” capacitors, they appear to have been discontinued. Instead, I have listed Elna “tonerex” as replacements, although I have never tried them. As for coupling capacitors, I recommend polypropylene for reliability, although you may prefer to use the more expensive paper in oil

types. I have been put off by the number of these that I have had to replace in vintage equipment, but I am probably being unfair.

In many places, there are potentially high voltages during warmup, so the resistors must have the correct voltage ratings as well as power handling. The

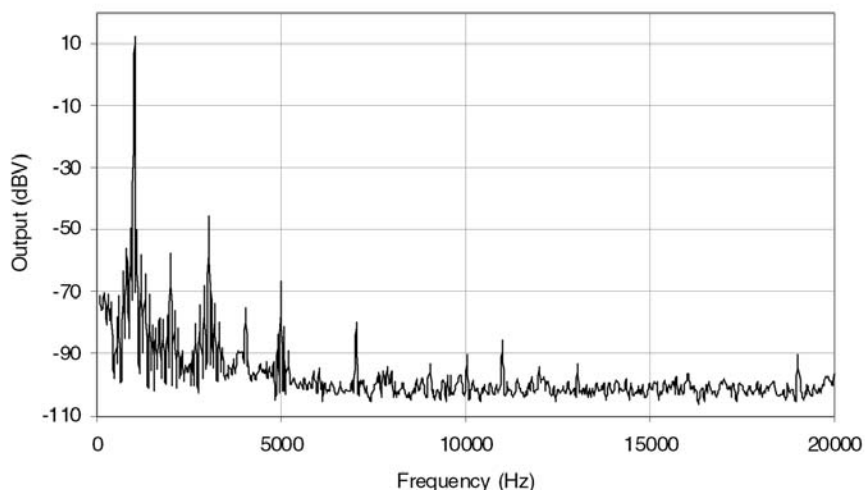


FIGURE 2: Harmonic distortion spectrum + noise without feedback due to 1kHz tone (2W into 8Ω).

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You will observe from **Photo 2** that the layout is a little cramped; I recommend using a larger chassis than the $12'' \times 9'' \times 3''$ one that I used. The amplifier produces quite a lot of heat and ideally the tubes should have more space around them for air to circulate. There

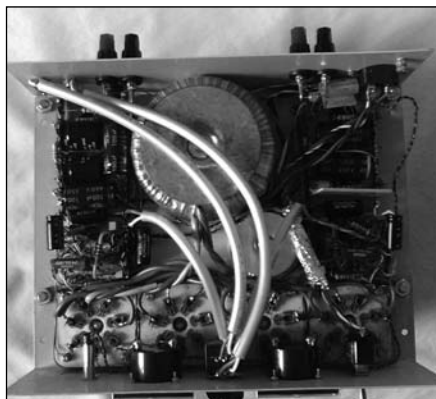


PHOTO 2: Underneath view.

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should also be good ventilation underneath the chassis.

While the retro style toggle switch on the front may look nice, the routing of the wires to it proved somewhat problematic. Notice the aluminum foil that I had to wrap around them to shield the sensitive input stage from the mains frequency radiation plus high frequency harmonics. This is exacerbated by the

fact that the current only flows during short pulses while the rectifier diodes are conducting. If possible, fit a rotary mains switch near the back with a long extension spindle or use a relay. I used tag strips throughout because this is a prototype and I knew from experience that design revisions would be likely. Connecting components directly to the tube sockets is generally a good

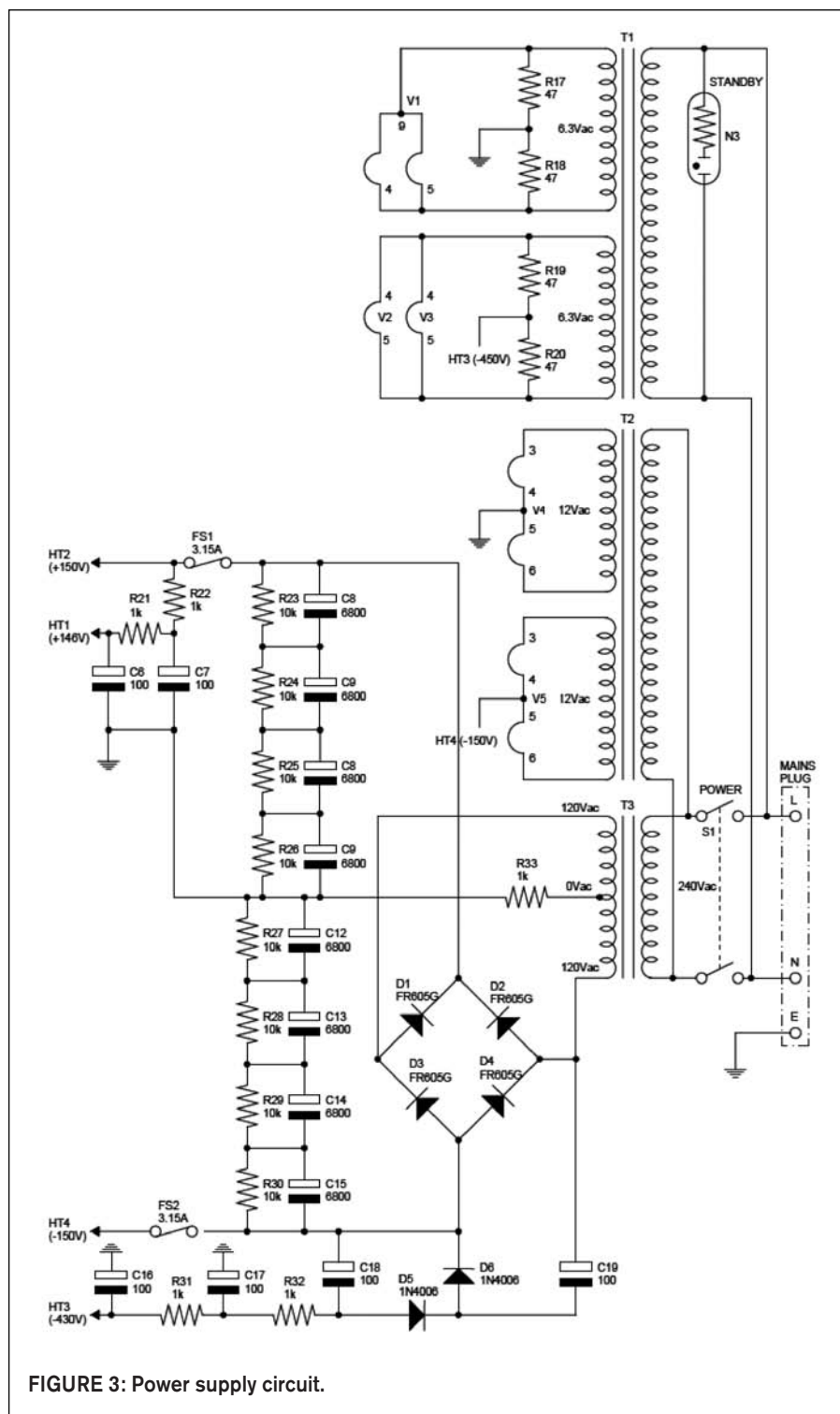


FIGURE 3: Power supply circuit.

idea because it keeps the signal paths short except in the case of the output tubes because the heat they produce will eventually destroy the component.

Before you switch on, check that the trimmer RV2 is in roughly mid position and that RV3 is set for *minimum* resistance. It is a good idea to arrange the connections so that this is the most anticlockwise setting. Then you can turn RV3 clockwise to increase the bias current from practically zero to the desired setting (I set it to 200mA), using the ammeter M1. During normal operation, M1 hardly twitches, so it is not a VU meter! However, it is reassuring to have it there as an early warning in case anything goes wrong.

Another good idea is to power up the amplifier the first time without any tubes plugged in just to check all the power supply voltages. After plugging in the tubes and setting the bias current, readjust RV3 if necessary after about 20 minutes. Then connect a millivoltmeter across the loudspeaker terminals and adjust RV2 to zero the reading. Always do this with the volume set to minimum or the input socket shorted.

When the amplifier is working, never switch it on again immediately after switching off or you will probably blow a fuse. Also, it is useful to have an aerosol can of contact cleaner handy because the tube pins and sockets tend to oxidize and will need a cleanup from time to time.

LINKS TO SUPPLIERS

www.cricklewoodelectronics.com/Cricklewood/home.php
<http://uk.rs-online.com/web/>
<http://uk.farnell.com/jsp/home/homepage.jsp>
www.maplin.co.uk/
www.chelmervalue.co.uk/index.htm
www.hammondsmfg.com/dwg20.htm
www.wollenweber-audio-modification.de/

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1. C. T. Hall, "Parallel Opposed Power Amplifiers" US Patent 2,705,265, June 7, 1951.
2. J. Futterman, "A Practical Commercial Output Transformer-less Amplifier," *J. Audio Eng. Soc.*, (1956 October).
3. Circlotron history page <http://circlotron.tripod.com/>.

ax

Table 1 Parts list

C1, C2.....Capacitor, 1µF 450V polypropylene Ansar	R12, R13, R15....Resistor, 100k 1% 2W 500V metal film Maplin
C3, C4.....Capacitor, 0.1µF 630V polypropylene Ansar	R14.....Resistor, 15k 5% 0.5W metal film
C5.....Capacitor, 10µF 250V electrolytic	R16.....Resistor, 10k 5% 0.5W carbon film
C6, C7, C18.....Capacitor, 100µF 250V electrolytic	R17-20.....Resistor, 47R 5% 0.5W carbon film
C8, C9, C10-15...Capacitor, 6800µF 63V electrolytic Elna "tonerex" or Samwha "for audio"	R21, R22.....Resistor, 1k 5% 0.5W carbon film
C16, C17, C19.....Capacitor, 100µF 500V electrolytic	R23-30.....Resistor, 10k 5% 0.5W carbon film
D1, D2, D3, D4...Diode (fast recovery), FR605G 6A 600V	R31, R32.....Resistor, 1k 5% 1W carbon film
D5, D6.....Diode, 1N4006 1A 800V	R33.....Resistor, 1k 5% 10W wire wound Welwyn
FS1, FS2.....Fuse and holder, 3.15A 20mm	RV1.....Resistor, variable 100k
M1.....Ammeter, 0-1A DC	RV2.....Resistor, trimmer 1k 20-turn 1W cermet Spectrol + 32mm panel mount adaptor
N1.....Neon lamp, wire ended, T2	RV3.....Resistor, trimmer 10k 20-turn 1W cermet Spectrol + 32mm panel mount adaptor
N2.....Gas discharge tube (GDT), 90V DC sparkover	S1.....Switch, double pole single throw 250V AC 5A
N3.....Neon indicator, panel mounted	SK1.....Socket, phono
PL1.....Plug, IEC chassis	SK2.....Terminals (shrouded) to suit loudspeaker cable
R1, R2.....Resistor, 34k 0.1% 0.25W precision metal film Welwyn	T1.....Mains transformer, 6V + 6V 15VA
R3, R4.....Resistor, 1M 0.1% 0.25W precision metal film Welwyn	T2.....Mains transformer, 12V + 12V 225VA
R5, R6.....Resistor, 100k 0.1% 0.25W precision metal film Welwyn	T3.....Mains transformer, 120V + 120V 625VA
R7.....Resistor, 470k 1% 2W 500V metal film Maplin	V1.....Tube, ECC83 + B9A socket
R8, R9.....Resistor, 4M7 5% 0.5W 3.5kV metal film Vishay (match pairs to within 1%)	V2, V3.....Tube, EF86 (matched pair) + B9A socket
R10, R11.....Resistor, 1M 1% 2W 500V metal film Maplin	V4, V5.....Tube, 6C33C (matched pair) + socket Chelmer
	Chassis.....Steel, 17" × 10" × 3" Hammond

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by Howard Ferstler



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List price: \$1990 (per pair)

HiVi, a leading Chinese manufacturer of audio systems, established research and design facilities in Toronto in 1994, and in 1997 merged with Swans Speaker Systems, opening corporate offices in Monterey Park, Calif. The company's huge product line runs the gamut from borderline minispeakers to towering, megabuck line-source models, as well as subwoofers. The M3 system is Swans' latest smaller model, and its performance should be indicative of what the company's larger systems can do in terms of spectral-balance issues and voicing.

The M3 is unusual in that speakers in

this size category (10" W x 12.5" D x 17.5" H) are normally two-way budget models and not three-way models with a luxury finish, upscale drivers, and, to quote the company's spec sheet on their Internet site, aspirations to become a "timeless classic." I do not know whether the system will become a timeless classic in the Klipschorn, AR-1, AR-3, Altec 604, original Advent, NHT Super One, Quad ESL 57, and Bose 901 traditions, but it is in any case a very good speaker system.

which I assume is also the diameter of the voice coil.) The cone itself is formed of Kevlar, supposedly to improve low-frequency dynamics. Whatever, it sure looks impressive.

The crossover network is fully fourth order, which means very steep slopes. The company rep told me that the woofer/mid transition point is at 800Hz and the mid/tweeter point is at 4.5kHz, and I confirmed this by doing some close-up driver curves. One advantage of 24dB slopes is

“ [The M3 has] perhaps the best-finished cabinet I have ever seen, with virtually no cosmetic flaws with either the box or the drivers. ”

FEATURES

The M3 is a vented-box design with a 6 x 2.5" flared port on the backside. The tweeter is a short ribbon job with a radiation surface that is about 2" high and 1" wide, although the width of the mouth in front of the tweeter is about 1.25", which would mainly determine its horizontal radiation pattern. The midrange is a 2" soft dome with a proprietary coating on the surface. The tweeter and midrange are mounted on a single plate, allowing them to be located closer together—a good idea that is also used with the company's floor-standing RM600R model.

The 6.5" woofer includes a phase plug in the middle to improve dispersion. (The plug is about 1.12" in diameter,

that they do a good job of protecting the midrange and tweeter drivers from out-of-bandwidth artifacts at the low ends of their operating ranges. They also minimize potentially audible anomalies such as lobing that result from two different kinds of drivers needing to deal with the same frequencies on either side of the crossover point. It is difficult to isolate all of the audible consequences of this particular design approach, but the manufacturer obviously sees advantages.

The spec sheet says the speaker is a "bookshelf" design, but it would require an unusual bookshelf to make this workable. The port is on the backside and the speaker cabinet is obviously designed to be oriented vertically. Few bookshelves, even those on top of a bookcase, would



happily accommodate the arrangement, because bookshelf placement normally requires a location close to the front wall. For the port to perform happily, the speaker needs to be some distance from shelf and wall surfaces, which means that this speaker would work best if installed on a stand and pulled out from the wall a foot or so.

The M3 is rated at a 4Ω nominal impedance, and the bandwidth is said to be 40Hz to 20kHz (no tolerances specified), with rated harmonic distortion of less than 1% down to 65Hz, with a 2.83V input at a 1m distance. Sensitivity is rated at 88dB, so the speaker is not playing particularly loud with the distortion being so low. The speaker is recommended for use with amplifiers rated at anywhere from 10 to 120W. (A plaque on the backside of the enclosure rates it at 10 to 70W.)

The cabinet is mostly solid wood, with an extra layer of MDF on the front panel, making it over 1" thick, and the exterior fit and finish are first rate. Indeed, I believe that it is perhaps the best-finished cabinet I have ever seen, with virtually no cosmetic flaws with either the box or the drivers. The surface is utterly smooth (but with decorative grooves engraved for aesthetics) and the front panel has curved edges that blend nicely into the side panels.

The speaker weighs in at a reasonable 26.4 lbs. The grille screen is easily removed and held in place by press-fit grille guides. The grille frame is moderately thick and not beveled or rounded on the inside, and those who favor listening to their speakers from up close may prefer to remove it to minimize diffraction artifacts that might offend—if only in their imaginations. Those who listen a bit further away in the reverberant field will not have any problems with the grille being left in place. The midrange driver's soft fabric dome has no protective screen over it, so I advise those who choose to remove and replace the screen regularly to be careful.

The cabinet back has two pairs of robust-looking wire connectors, with flat straps between each pair that you can remove to bi-amp the speakers. (Both the connectors

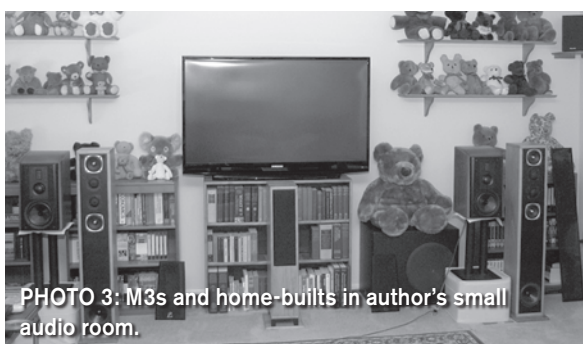


PHOTO 3: M3s and home-buils in author's small audio room.

and the straps appear to be gold plated.) The positive and negative posts are spaced about 1.75" apart (a legal requirement in Europe), which means that although they can handle bare wires, pins, spades, and single banana plugs, they cannot handle GR-type double bananas.

MEASUREMENTS

Before listening to the speakers, I decided to do some basic measuring. However, there appears to be substantial contro-

versy regarding which measuring techniques you can use with loudspeakers that really highlight the audible consequences.

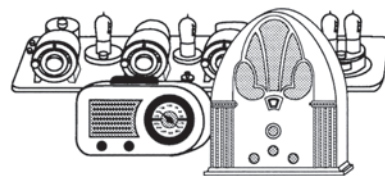
Certainly, it is a good idea to determine a speaker's bandwidth, and, of course, it is also important to determine whether a speaker has any obvious distortion artifacts such as buzzing, low-bass doubling, or huge cancellation or driver-output notch-

es—not to mention overlap bumps—at or near the crossover regions. More esoteric measurements that would be helpful during the design process, but of questionable value when testing the final product, might involve impulse response, step response, energy-time analysis, square-wave behavior, waterfall plots to determine driver or network energy storage, and even coil/gap analysis.

And, of course, there are PC computerized, gated (pseudo-anechoic) tests

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that allow individuals to measure the direct-field output of a system at assorted angles (horizontal and vertical) around the enclosure. Doing the latter would require maybe hundreds of such plots at hundreds of angles to get a workable map of the output. And decoding the results would require a lot of brainwork, but this seems to *not* bother some people in the least. One reason may be that few go to the trouble of doing the required multitude of measurements.

Rather, some do a few response-curve measurements at a 1 m distance, often on- and off-axis a few degrees (usually just horizontally), and then assume that this approach will determine just what a speaker will do when listened to at typical—and obviously much greater—distances in typical rooms. The problem is that the further you get from the speakers, the more the multitude of direct-field outputs at various angles tend to merge into a whole that defies being analyzed by the 1 m approach. The balance between the direct-field and reverberant-field outputs begins to tilt in favor of the latter, at least at most frequencies below the sometimes beamy treble.

I might point out that some reviewers obtain assorted test programs that can be downloaded to their computers, and then use them to deliver graphs aplenty—but do so without showing how the various plots relate to what you hear with speakers playing at typical listening distances in typical rooms. The idea seems to be to bury the reader in arcane data (blinding them with science, so to speak) and then let them come to their own conclusions. Interestingly, the reviewer's supposedly objective conclusions may ultimately depend upon single-presenta-

tion listening sessions that seem completely unrelated to what was measured.

My approach involves using some basic time-honored techniques. I place the speaker pairs in different spots that work well with my normal, couch-located listening positions and then do a series of 20-second, moving-microphone readouts, ranging left-to-right and back, over a roughly 1 × 1 × 5' horizontal area near seated ear height. While this is happening, my AudioControl SA3051 real-time analyzer is operating in its 20-second averaging mode, with the 1/3-octave readout then summarized and put into storage. I do this up to five times with the device, and then average the five averages themselves into a 100-second summary.

Generally, each of the five will be done with the speakers in somewhat different "ideal" locations, with both forward-facing and toed-in arrangements involved. The source material will always be stereo (uncorrelated) pink noise, which allows for the most homogenized, stereo-soundfield result. I believe this procedure reasonably determines the ability of the speakers to deliver a smooth power input to the room. The area I normally use is typically furnished, and is a bit more than 22 × 18' in size, with an 8.5' ceiling. I position the speakers in front of the longer wall, minimizing close-up sidewall- and corner-related artifacts.

Now, of course, this procedure does not measure radiation patterns and distortion levels, but it does give you an idea of bandwidth and spectral-balance smoothness, and if there are any energy storage artifacts involved, the plot should show those of audible significance.

My listening experiences continue to

tell me that for practical purposes this procedure does a pretty good job of determining the meaningful response smoothness of speakers in typical listening rooms. *My confidence in this approach is high, since I have had so many speakers that measure similarly also sound similar during careful, level-matched listening comparisons, and vice versa.*

The response curve (**Fig. 1**) shows two readouts. The solid line includes a Velodyne F1800RII subwoofer handling the bass below 90Hz. The dotted line at the bass end shows the M3 bass driver's unassisted reach. Two of the five 20-second plot runs were done with the speakers elevated 18", and the three others were done with the speakers elevated 28". Two of the five plot runs had the speakers toed in; the other three had them facing forward. With each of the five plots the speakers were at varying distances from each other and from the front wall. As you might expect, these varying locations and aim angles only impacted the bass and treble performance. The midrange performance was consistent with all five.

It is easy to see that with my 100-second averaging technique the speakers are fairly smooth, although the overall response tilts downward considerably toward the treble. While this looks like the speakers might sound dull, this kind of slope can work well with many classical recordings and also keeps a lot of pop music from sounding more brittle than it should.

There are some artifacts in the curve that need explanation. The dip centered at 315Hz is the result of quarter-wave-length cancellation artifacts (sometimes called the "Allison Boundary Effect," in honor of Roy Allison, who pointed out the phenomenon) that happen when reflected energy in the bass and midbass cause interference cancellations with the direct radiation from the bass-range driver or drivers. The notch location would vary slightly, depending upon woofer distances to wall and floor surfaces. This is not a speaker anomaly, per se, but always occurs with speakers located at a finite distance from room

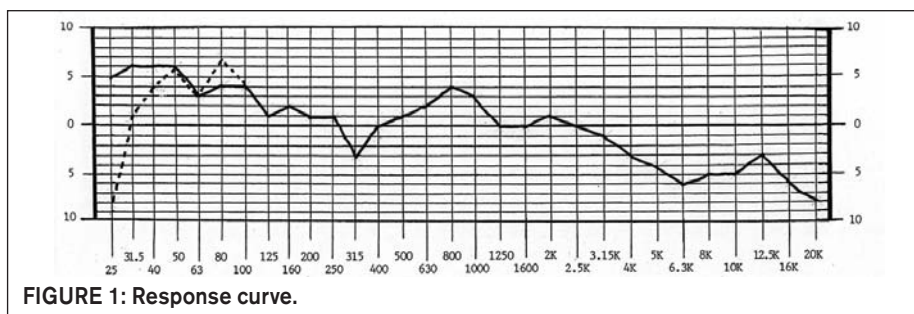


FIGURE 1: Response curve.

boundaries. The easiest way to minimize the artifact is to locate the speakers so that the three closest boundaries are different distances from the woofer center. If you cannot do this, standard equalization procedures should do the trick, provided the dip is not too deep, which is the case here.

A second artifact is the broad peak centered at 800Hz. I am not sure what is causing this, but because the crossover point between the woofer and midrange driver is at this frequency, it may be related to crossover design. Reversing the midrange driver polarity might flatten things a bit, and I had intended to try this after I was done with my measuring and listening work. However, the driver/wire hookups inside the box are actually made with solder instead of push-tab connectors, a not uncommon practice with "serious" speakers these days. While I sometimes like to diddle with speakers being reviewed, I decided not to mess with a good construction job.

LISTENING TESTS

After completing the measuring work, I located the speakers for best performance, and then listened to see whether I could hear any of the anomalies (or lack of anomalies) I measured. Generally, I use the same batch of recordings with each review to keep things consistent.

I then set up a second pair of speakers as reference models and did some level-matched A/B comparisons. When I do this kind of procedure, I locate the two pairs with roughly equal spacing between each (this means orienting them AB/AB), so that the soundstage widths (and resulting sense of left-right spaciousness) will be similar. Comparisons of this kind mandate using two separate stereo amplifiers with at least one of them having level controls. To best level-match, I adjust the gain of one pair so that the 20-second averaged RTA curves overlay each other as much as possible.

In the old days, I used a pair of Dunlavy Cantata speakers as my reference units when doing A/B comparisons (list price at that time was \$5500 per pair),

and listened in my smaller audio room. Their size and shape allowed most other speakers to be arranged close to them, with minimal problems regarding speaker-to-speaker interactions. I sold the Dunlavy units, and now have some home-built, 4' tall speakers that use Allison tweeters and Tang Band 4" midrange drivers in a vertical MTTM array. A 6.5" Allison woofer at the front-panel bottom handles the bass, with good response, down to about 65Hz. For serious (non product-reviewing) listening in that room, a Hsu VTF-3 (MK) subwoofer handles the low end below 90Hz. However, I used neither the sub in my main room nor the one in my smaller room with any of the M3 listening comparisons.

My home-built speakers are quite good (and might sell for \$2000 a pair if they were marketed), and while this paragraph and the one that follows looks like a review inside of a review, in an A/B faceoff against the Cantatas before I sold them the result was a spectral-balance and transparency draw. Even in terms of central imaging they were equal to the Dunlavy units, although nobody is going to say that their 6.5" woofers are comparable to the 10" jobs in the Cantatas, if low-bass performance is critical.

When I compared those home-built units to the even larger Allison IC-20 units in my main system (their list price back in 1991 was \$5200 per pair; these days they might sell for \$8000), it was often difficult to tell the two pairs apart in terms of clarity and spectral balance, at least above the low-bass range. Admittedly, the angled-panel IC-20s sounded more spacious than my home-built units. Clarity was essentially identical, and those home-built units actually had tighter sweet-spot center imaging.

With the main-room comparisons, the IC-20s were positioned up against the front wall (where the company indicated they should go) with the M3 units out in front of them, 28" from the floor and 2' from the front wall. Of course, they were also shifted somewhat to the sides to not block the IC-20 units.

When comparing with my home-built systems, I placed both pairs side-by-side in both the main room and later in my smaller room, several feet out from the front wall, and as with the other comparison, located AB/AB style to keep the soundstage widths the same.

My main source material was a CD compilation engineered by the late John Eargle: *Engineer's Choice* (Delos DE3512). This is a great comparison disc, because it has excerpts from many musical works by many composers, with different instrumental types and recording techniques. In addition, I used a wonderful recording of Handel's *Twelve Concerti Grossi*, Op. 6 (Denon 81757-6305) and an equally wonderful recording of Vivaldi's *Le Quattro Stagioni* (Opus 111 30363). All three have enough in the way of clarity, nuances, detail, smooth sound staging, and just plain realism to satisfy just about any speaker reviewer.

In addition, I also checked out the speakers with a fine rock-music recording, Dire Straits *On Every Street* (Warner 26680). The comparisons were revealing, because the Swans did exhibit some notable differences from my two similar-sounding reference systems.

With most of the selections the violins and guitars sounded almost identical in terms of spectral balance with all three pairs. This indicates a smooth treble range with the M3, although at times the instruments did not have quite the high-range shimmer and strength of my IC-20 and home-built models. With medium- and large-scale instrument works (Shostakovich, Bizet, Bach, Vivaldi, Copland, Handel, and so on), the M3 units at times sounded fuller, but also maybe a bit thick or "chesty," with that 800Hz rise allowing the speakers to sound more forward than the reference models and maybe a tad on the "boxy" side. Much of the time the reference models sounded a bit more transparent, too, with better treble delineation. The flatter measured extension into the treble range by the reference models would account for this. In virtually all cases my IC-20 systems sounded more spacious

than my home-built systems, which sounded more spacious than the M3s.

Admittedly, spaciousness involves taste, and some listeners prefer that their speakers de-emphasize that characteristic, because it allows the room to dictate more than they would like. Spaciousness, in any case, seems to be strongly influenced by the dispersion characteristics in the upper midrange and treble, and at those frequencies the IC-20 systems, in particular, outclass just about any other forward-facing speakers available.

With smaller-scale materials (Mozart quartets, Warlock guitar) the differences were often quite small, but with that 800Hz peak again sometimes intruding into the behavior of the M3 units. A female vocal selection had the M3 units pulling to a draw against the references, but with the M3s sounding a bit more forward.

I am one of those people who do not

consider rock music all that effective as a test tool, because there is often no acoustic-space, live-music experience to use as a reference. Virtually all of the recordings that are any good are studio jobs, with enough electrical amplification to nullify any sense of concert-hall soundstaging. Nevertheless, in some ways the style can at least highlight punch, bandwidth, and clarity. With the Dire Straits material the M3 pair managed to do quite well, with maybe a bit more rock-grade bass punch than either of the reference pairs, but with less ultimate bass extension than the IC-20s.

I also did some test-tone bass sweeps with the M3 units to make sure that serious bass artifacts were not there. The speakers were impressive, considering their size, and I got solid results down to 35Hz, with decent performance down to 30, although at a subdued level.

With some ported (and even un-port-

ed) systems you begin to hear troublesome artifacts as a sweep heads toward 25-20Hz, but with the M3 units the results were impressively clean. Not much output down there, but no weird doubling artifacts, either. Those are good little woofers. Still, as the printed curve shows, the speakers are small enough to warrant some assistance from a good subwoofer if you require really powerful and deep bass from your installation.

So, how do I like these speakers? Well, at \$995 each these are very attractive, well-designed, and good-sounding products. A careful search may find a few lower- and equal-priced speakers that do slightly better on the technical metrics many believe are the keys to good loudspeaker sound. However, the M3 units hold their own with any of them when you factor in build quality, low-bass reach, and appearance. Thus, I recommend them. **rr**

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Dayton T1503A Subwoofer

by Tom Nousaine



Dayton Audio T1503A Titanic Subwoofer
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This interesting product actually turns out to be a real subwoofer. By that, I mean it actually produces sound pressure from 12-20Hz. Most of what manufacturers call a subwoofer these days is simply a woofer that runs out of gas by 25Hz. Many of them also have a dynamic (SPL) capability that peaks at 50Hz or 62Hz and then falls by 12-20dB per octave as frequency falls.

None of that is true with the Dayton T1503A, which will produce a true 81dB of SPL (relative to 2m, a real possible listening distance) at 12Hz and 105dB SPL

at 25Hz. Of the dozens of commercial "subwoofers" I've tested, this is only the third one I've found that is capable of the title.

to prevent two of the four plastic snap-ins from being broken during shipment. The cabinet also had a small amount of exterior shipping damage at one corner. I recom-

"It's kind of like getting a date with Charlize Theron and finding her lipstick already smeared when you pick her up."

FEATURES

This sealed box system contains a 15" driver and a claimed 1000W amplifier with a single band parametric 18 to 80Hz equalizer. The 19 3/8" depth includes the front panel cloth grille but fails to include the 1/2" depth of the rear panel controls. The enclosure appears to be painted particleboard that has a definite hollow sound when knuckle-rapped.

However, because the system is a subwoofer and the relatively small cubical enclosure is highly resistant to ballooning, the box must be just stiff "enough" and not in need of the excessive weight and damping sometimes seen in commercial and DIY products. While there's nothing wrong with an enclosure that is stiffer or heavier than it needs to be, someone must pay for the materials and labor. And that is usually the customer. Suffice it to say I wasn't able to hear any cabinet sound when listening to the product.

The grille is nicely done but the packaging for shipment wasn't sturdy enough

mend that Parts Express spiff up the packaging. It's kind of like getting a date with Charlize Theron and finding her lipstick already smeared when you pick her up.

Performance

The following chart describes the frequency response (taken at 0.5m ground plane) and actual response of the crossover at marked points.

Frequency Response

LFE Input: 36Hz to 219Hz ± 2.2 dB

Stereo Input: 34Hz to 189Hz ± 2.5 dB

Low Pass Characteristic

Low Pass Dial Marking	True Low Pass Turnover Frequency	Level Change
LFE	219Hz	-
200Hz	189Hz	-
3 pm	164Hz	-
105Hz	111Hz	-
9 am	56Hz	< 1dB
30Hz	46Hz	-9dB

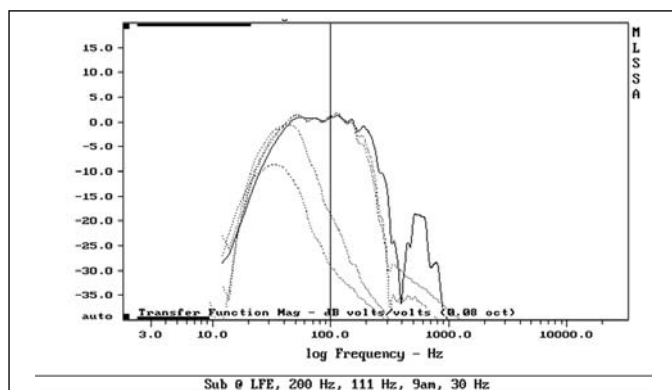


FIGURE 1: Crossover control action from LFE (solid) and all subsequent positions counterclockwise.

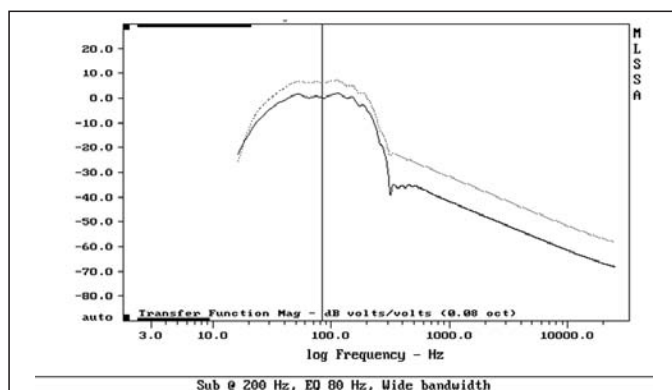


FIGURE 2: Equalizer with wideband 1.0 setting simply works like a level control.

TABLE 1: EQ PERFORMANCE

Setting	Frequency	Bandwidth	Level
EQ80 1.0 +6dB	Full Band	Full Band	+5.8dB
EQ80 0.1 +6dB	78Hz	55-99Hz	+5dB
EQ80 0.1 -14.5dB	83Hz	48-117Hz	-12.9dB
EQ73 0.1 -14.5dB	64Hz	46-92Hz	-10.9dB -1.5dB
EQ60 0.1 -14.5dB	62Hz	44-80Hz	-9.7dB -3.7dB
EQ53 1.0 +6dB	Full Band	Full Band	+ 5.6dB
EQ53 0.1 -14.5dB	55Hz		-7.6dB
EQ53 1.0 -14.5dB	Full Band	Full Band	-13dB
EQ48 0.1 -14.5dB	48Hz		-7.0dB

This is very good for a subwoofer low pass. Typically there is little relationship to dial markings and actual low-pass frequency, and often there is significant level/crossover control interaction. With the T1503A, you can use the controls with confidence without needing to always rebalance levels when changing crossover frequency.

Table 1 shows measured performance of the equalizer at various settings of frequency (EQ80 to EQ18), bandwidth (0.1 narrow or 1.0 full wide) and level (+6dB or -14.5dB). The band-

width and level measurements are all simply the maximal settings. In general at narrow bandwidth settings the equalizer true frequency is close to the dial markings, but the higher level control settings simply work like a full bandwidth volume control.

TABLE 2: DYNAMIC PERFORMANCE

Frequency	SPL
62Hz	104.8dB
50Hz	105.5dB
40Hz	102.6dB
32Hz	104dB
25Hz	104.9dB
20Hz	96.7dB
16Hz	86.3dB
12.5Hz	80.8dB
Average	104.4dB 25-62Hz
Uniformity	98.9%

Manufacturer specifications seem to always overstate dynamic capability. For example, Dayton specifies the T1503A at 115dB SPL with no frequency specified, but this product isn't capable of

“This product is fantastic in two ways that make it preferable to most commercial speakers.”

For example, setting the frequency to 80Hz and level to +6dB with a 1.0 bandwidth simply raises the full band by 5.8dB. Also, setting the equalizer frequency to 18Hz through 32Hz has little effect other than cutting the level below 66-75Hz, depending on dial frequency. In other words, at settings from 18 to 32Hz the equalizer simply has no effect when set to gain and simply sharpens the nominal high-pass slope when set to cut. There is also occasionally an overall level interaction with certain EQ settings, meaning that you may need to rebalance the level at certain EQ settings.

This performance is typical of single band EQs on active subwoofers although the Dayton seems to work more closely to marked settings than many others. However, it will still be difficult to use the equalizer except in rooms with an obvious single mode problem at a relatively high frequency.

Table 2 depicts the actual sound pressure reproduced by the subwoofer at 1/3 octave frequencies using the Linkwitz/Keele 6.5 cycle ramped sine waves at 2m in an optimal listening position in a 7600ft³ great room. In a smaller 2100ft³ room 2-3dB more SPL was generated by identical subwoofers in a previous experiment. This measure tells you what kind of actual dynamic performance to expect.

generating 115dB even at 1m listening distance at any given frequency.

That said, this product is fantastic in two ways that make it preferable to most commercial subwoofers. As mentioned before, it is a real subwoofer capable of reproducing real sound pressure as low as 12Hz. But, even better, the dynamic capability is fantastically even. For example, it averages 104dB SPL from 25 to 62Hz with nearly perfect uniformity. Most commercial models tend to have maximal SPL at 50 or 62Hz and then fall off at 12-24dB per octave at lower frequencies. This tends to give them a slight advantage in my 25-62Hz SPL “Heart of Bass” average.

That's why I invented the “Bandwidth Uniformity” statistic so a “subwoofer” that can do 110dB at 62Hz but only 98dB at 25Hz, which gives a 104dB SPL average 25-62Hz, only gets a 94.6% Bandwidth Uniformity score while a subwoofer, like this one, that can produce 104dB at practically every frequency gets a 98.9% score. However, the T1503A would occasionally “pop” and “fart” at 12.5 and 16Hz when overdriven, so users should exercise care when using new program material that has super low frequency content.

So, the T1503A is a true subwoofer



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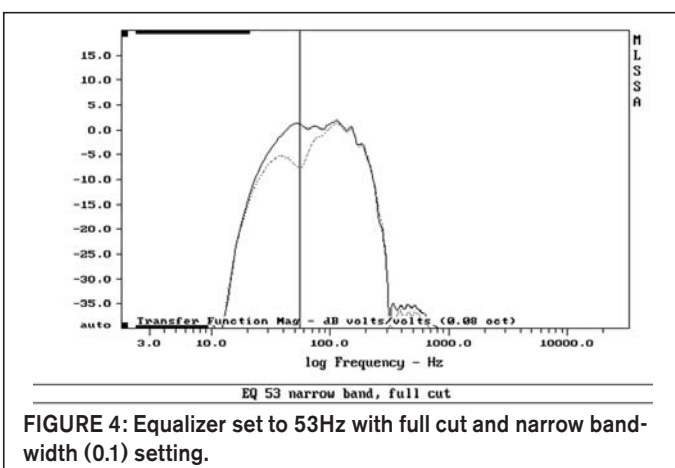
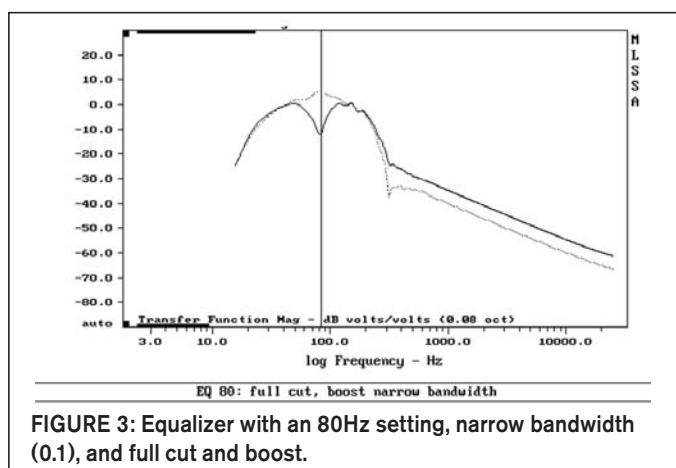
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with wonderfully uniform dynamic capability. Bravo!

Setup and Listening

Setup is mostly straightforward for me because I've already response-mapped my room for subwoofer position with MLS-SA. So I just left it in the optimal corner position in this room. Because the crossover has no level interaction and I already knew that the crossover control markings were reasonably accurate in the middle of the dial and didn't have any level interaction, it was easy just to dial in 111Hz and set the level to 80dB with my sound level meter. This matched up with my personal subwoofer (eight 15" TC Sounds drivers in an 8' infinite baffle built into the floor), and with a dB Sound switch box I was able to switch directly between the two systems for a casual listening session.

Just for clarification, I use the 120Hz low-pass filter on my Anthem processor. Why so high? Well, my subwoofer system has a natural low-pass function that begins at 70Hz, meaning that it begins rolling off acoustically at 70Hz relative to the listening position. So 120Hz electronically is a natural. This also let me match the Dayton with its internal crossover to the system.

How about listening? Well, I find it fascinating that many folks seem to think there will be a difference in performance and sound quality between movie and music programs. This is just plain silly. A good subwoofer will perform equally well no matter what the program choices. Sure, if you only watch drama and listen only to

chamber music you may not need a "subwoofer," but that's a completely different question. It might be useful to distinguish between bass-head low-frequency music versus regular rock, which has no real deep bass, but again that's a different question.

The truth is the frequency content matters and not the artistic venue. There are now plenty of movies—as well as plenty of compact discs—with deep bass. That seems to be a result of two factors: First, the advent of digital recording equipment means that true subsonic sound can be captured. Second, the advent of the THX Cinema with active subwoofers that have their own electronic high-pass filters means that soundtracks no longer need to be high-passed at 25Hz or even higher.

In fact, recordings of nature often have the deepest bass. The thunderstorm has really low deep bass. And recordings of locomotives and aircraft also can have ground-shaking, deep bass. So if you watch/listen to adventure movies, organ music, and nature recordings, you need a subwoofer, but the one you choose will function equally well with anything you put through it.

WOOFER MYTHS

Also, there's another myth relating to subwoofer driver size: the idea that an 8 or 10" "subwoofer" (there's an oxymoron) will have better "punch and speed" than a 12, 15, or 18" model. The truth is that an 8 or 10 will just have less low bass. Think about musical instrument speakers, for example. They just prove that a 15 or 18" speaker can reproduce mid and up-

per bass and midrange just fine.

For example, when I first built my custom subwoofer, I experienced an interesting problem, mid-bass compression. That is, the subwoofer (122dB SPL from 12-62Hz <= 10% distortion) had way more dynamic capability between 60 and 100Hz than my left and right upper channel speakers each with a pair of 6.5" woof-

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ers. In other words, at high output levels my main speakers couldn't keep up with the subwoofer in the mid-upper bass region. Or, it could be said that my main speakers weren't *fast enough* dynamically to keep up with the subwoofer. The solution was to add an active bandpass 10" "subwoofer" to the left and right main speakers that function between 45 and 90Hz.

The myth of woofer speed is also just silly. The "speed" of sound at any given frequency is a function of the frequency. If a speaker can reproduce the frequency, it is simply "fast" enough. If it were too slow, it couldn't do the frequency. What happens is that if the speaker lacks the displacement to produce the frequency loudly enough, then to some people it may sound "fast," but what it really means is that it doesn't have sufficient bass for the program.

While it is true that high-quality low-frequency drivers often have heavy cones, it is also equally true that they have heavy frames and heavy magnets which have more energy than those in a "fast" midrange or tweeter. So the idea of fast woofers is just a myth, and the systems that sound slow and heavy will simply be a function of sub-optimal installation and setup.

If "speed" was an issue, then the Dayton should be faster than my eight 15" system. But, on the other hand, was it fair to compare a single 15 with a system eight times bigger? Well, why not? Shouldn't a single 15 walk circles around a big system on acoustic bass?

The truth of the matter was that on material with acoustic bass the systems sounded essentially the same. It was true that on oc-

casion the big system had obviously more moving air than the Dayton, and you could argue that it was too much. Remind yourself that the air pressure, while it may have been more than that you'd receive in a club with an acoustic bass with either of these subwoofers, it didn't lack in frequency definition, instrument position, or clarity. The Dayton sounded just as good and arguably slightly more natural. But I could have made them sound exactly the same with a lower high-pass filter on the big system.

Where it lacked equality was with CDs with super deep bass. On the Bass Connection bass-head "Pure and Perfect Bass," the downward sweep to 5Hz was not as strong and floor-shaking with the Dayton. But the Dayton was impressive nonetheless, reproducing sound that the high-end system with four 12" woofers or the one with a 12 and 18 driver wouldn't even attempt. That's because the former was a 19Hz bass reflex where the system falloff below 18Hz would be 24dB per octave even if it weren't also electronically high-passed. And the other system was electronically high-passed at 15Hz.

This is a real subwoofer and at \$899.95 must be considered the best bargain on the market for those who want a true subwoofer. True meaning it doesn't give up at 20 or 25Hz, and properly set up will sound great with any program material. The operating controls are functional enough to allow proper setup, but using the equalizer will be much easier if measurement equipment is available.

Although my unit came fully assembled, the "preassembled" option saves you \$150

and is a simple screw and strip method. That is, you must strip the speaker leads and need only a utility knife, a $\frac{5}{32}$ " hex driver, a wire stripper, and a screwdriver. No solder or other equipment is necessary. It appears that you could finish assembly—including unpacking and storing packaging material—inside an hour. So the DIY option is a great value, unless, of course, you know somewhere else you can get more than \$150 for an hour of your labor.

Manufacturer's Response:

Thank you for reviewing the Dayton Audio T1503A Subwoofer. We feel the review addressed many of the attributes of what really makes a subwoofer a true performer. We also feel the review was a fair analysis of the T1503A and supports our belief that the subwoofer is an exceptional value for anyone looking for a new sub. As mentioned in the article, it is also available in kit form, thereby saving the consumer money while providing an excellent "I did it myself" experience. Though many instances of shipping damage come from mishandling, we are working to improve our packaging in order to prevent any shipping damage to the Titanic Subwoofers in the future.

Rich Taylor
Marketing Manager
Dayton Audio

rr

Authorized retailers of Dayton Audio components include

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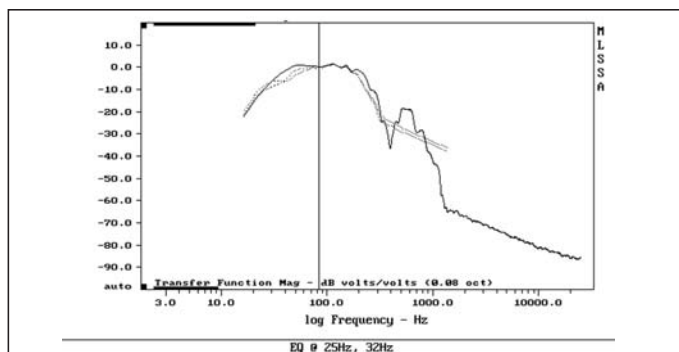


FIGURE 5: With the equalizer set to 25 and 32Hz cuts simply work more like high-pass filtering.

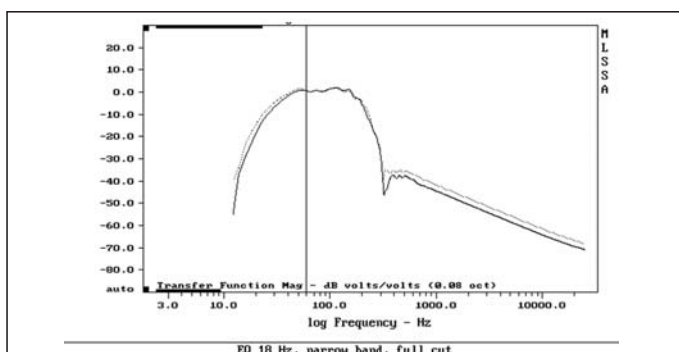


FIGURE 6: With the 18Hz setting the equalizer has little functional action.

CORRECTION

In the December 2009 issue of *audioXpress*, reviewer Tom Krehbiel's last name was misspelled. We regret the error.—Eds.

COMPONENT RELIABILITY

I very much appreciate reading Ed Simon's "Distortion Meter" article (Nov. '09, p. 22). At first I resisted reading it because I do not see myself building the test equipment even though I have made and repaired many electronic projects. I liked the "Lessons Learned" part of the article because they agree with my feelings and experiences, and, because they were derived from the measurements from test instruments, I can accept them as fact as opposed to opinion. This year I have done projects in which I have replaced every electrolytic capacitor in a power amplifier (29 years old) and a CD player (15 years old).

I have made it a standard practice to clean the leads of each capacitor with metal polish to remove the oxide layer and to make the solder joints more reliable. I find it hard to rely on the combination of rosin flux and the temperature of melted solder to break down this oxide barrier. This is especially true when you do not want to damage the component with overheating or cause the copper tracks on the PCB to lift off.

Once, a solder joint looked good under a magnifying glass and had functioned properly for 15 years, then became a high resistance connection. All it took to correct was a touch of the soldering iron to remelt the connection to make it low resistance again. The value of gold is that it does not oxidize, but the plating is so thin that several insertions may wear it away. This is especially true with cheaper connectors, and then performance highly depends on the metal underneath the gold plating.

I also enjoyed the April article on modifying the Music Hall CD25.2 by Kit Ryan (p. 6). The author noted several instances in which the theoretically optimum component was not the best-sounding. I wonder whether this occurred because of oxide in the connections. In testing, the author used

pin sockets to insert the leads of components so that he could quickly and easily swap components. Because the leads of the capacitors vary in diameter and in material, their connections can be highly variable. The thicker leads would have higher contact force. Copper leads have better conduction and solder better than steel leads, even if both are solder-tinned.

Another factor that comes to mind is that most high fidelity equipment reviewers allow the equipment to burn in for several days and I wonder whether this was the case with each component switch. I do not intend to question the integrity of the author, as by the time that one component is allowed to burn in, you would forget what the previous version sounded like and so the tester would need a minimum of two CD players to do a proper AB test. The first CD player would have component A in circuit and the second CD player would have component B in circuit. Ideally the

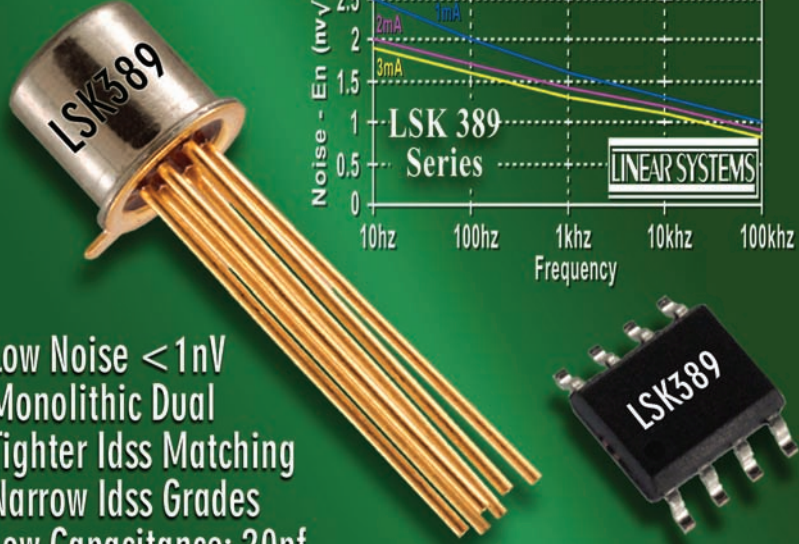
tester should have as many CD players as there are components to test. I know that this would be highly impractical and time consuming, but considering the small differences it may be the only way to reliably test changes until they make available test instruments to quantitatively measure image depth and width, bass dynamics, musicality, and other audiophile criteria.

*Fred Arikado
Toronto, ON, Canada*

Ed Simon responds:

Thank you very much for your comments. They are appreciated. As to the oxide on component leads, there is quite a lot to be said on that subject. Three important notes: Some gold plating is only cosmetic and the base metal can oxidize through the plating; steel in any audio lead may increase distortion; and, finally, cleaning the leads on parts before soldering them in is a good practice on parts that are a bit older than brand new.

1nV Low Noise Dual JFET



The image shows a cylindrical dual JFET component labeled LSK389 with four gold leads. To its right is a graph of Noise - En (nV/√Hz) versus Frequency (Hz) on a log scale from 10Hz to 100kHz. Three curves are shown for bias currents of 1mA (blue), 2mA (purple), and 3mA (yellow). All curves show a decrease in noise as frequency increases. The 1mA curve is the highest, followed by 2mA, then 3mA. The graph is labeled 'LSK 389 Series' and 'LINEAR SYSTEMS'.

Frequency (Hz)	1mA (nV/√Hz)	2mA (nV/√Hz)	3mA (nV/√Hz)
10	2.5	2.2	2.0
100	2.0	1.8	1.6
1k	1.5	1.3	1.2
10k	1.2	1.0	0.9
100k	1.0	0.8	0.7

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Kit Ryan responds:

Mr. Arikado raises so many imponderables in his letter that we could study them for ages. When examining circuitry, the number of variables becomes unmanageable very quickly. The trick is to decide which ones have the most influence on the characteristics of interest. I won't say I'm a great believer in the "breaking in" of purely electronic components, although I've seen something on how Teflon capacitors need 500 hours before sounding good. One problem is that I haven't listened to my system for 500 hours in the last four years—who's got the time?

Also, I've never seen anything in the vendor literature supporting this, unlike mechanical components such as speakers where break-in is clearly a real phenomenon. Overall, most of the components I tested had quite obvious differences, and they all had about the same amount of time to become acclimated, so I wouldn't expect a whole lot of difference over time. In my own system, I have now had the selected components in place for perhaps 200 hours of playing, and it sounds the same as it did during the early testing. Sure, no one's ears have absolute long-term acoustic memory, but using my very "broken in" LP system as a reference, the comparative CD player performance is about the same as it was. Therefore, I'm confident that break-in is not a significant factor in this assessment.

As to the "oxide" on the leads issue, because the components were inserted numerous times, their contacts "swiped" the sockets regularly in a short time. This pretty much ensures that a good contact was made. The gold-plated socket contacts were in good physical shape during the testing and seem to be performing up to par now, months later.

The idea that there is a "theoretical best" component pre-supposes that we actually know all the important parameters of components and how they relate to audio. I contend that we don't; otherwise, all the arguments would have been resolved long ago. For example, my system is full of tantalum coupling capacitors, generally despised throughout the audio world, yet the system blows away a very expensive tube system that I heard recently with its custom capacitors. There's more going on than we can yet measure, and I plan to spend a lot of my remaining years trying to find out why.

STROBE VALUE

Concerning the spirited exchange between Mr. Galo and Mr. Marovich about

the KAB Speed Strobe (Xpress Mail, Dec. '09, p. 42), and what with inflation and all, the KAB's price is in the ballpark of a good strobe and disc purchased back in the 60s.

*Mike Laird
Port Angeles, Wash.*

REDUCING JITTER

I have been a long-time subscriber to all of your magazines as long as I can remember. I have complete collections of all of the issues of *The Audio Amateur*, *Speaker Builder*, *Glass Audio*, and *audioXpress*. . . I greatly admire what you have created, and in my journey from fervent amateur to dedicated professional have greatly relied on the information in your publications.

I was therefore quite disappointed to read a serious error in Jan Didden's recent interview with Dr. Malcolm Hawksford in the November 2009 issue. On page 17, Mr. Didden asks: "Benchmark Media Systems claims that their DAC1 products succeed to almost get rid of jitter completely because they put a very clean clock next to the DAC with an option to slave the transport clock to it. Their USB interface apparently works the same in that it actually "requests" samples from the media player or PC, at a rate dictated by the clean DAC clock."

The information in this "question" is completely erroneous and misleading. It does a great disservice not only to the readers of the article, but to the engineers and companies who have actually accomplished what Mr. Didden mistakenly says that Benchmark has done.

The Benchmark DAC (www.benchmarkmedia.com) has achieved a good deal of success and offers a product that performs well for its relatively low asking price. They were one of the first companies to incorporate the Analog Device AD1896 Asynchronous Sample Rate Converter (ASRC) in their products as a way to reduce measured jitter.

All of the digital inputs to the Benchmark products are fed to the ASRC, where they are converted to a constant sample rate of 115kHz. The AD1896 has a very clever design that sharply reduces the amount of measured jitter in the incoming signal as the sample rate is changed.

The device is called "asynchronous"

because a jittery incoming signal is presented to the ASRC and the new data (completely interpolated from the original data) is clocked out at a more constant rate by the fixed-frequency, low-jitter master audio clock to the DAC chip.

This same technique is applied to data that is fed to the USB input as well. However, there is *no* communication to the media player or PC whatsoever by the Benchmark.

What Mr. Didden has described is instead a mode of USB audio transfer called "asynchronous." In this case, the "asynchronous" USB transfer mode has *nothing* whatsoever to do with the "asynchronous" sample rate converter employed by Benchmark. Instead, the "asynchronous" USB transfer mode means that the master audio clock is not synchronized to any of the clocks in the media player or PC.

With asynchronous clocks, there is a problem created as the data rate of the source will not match the data rate required by the DAC chip.

With ASRC, the problem is "solved" by throwing out *all* of the original data and interpolating new data that represents what the data would have been if a low-jitter clock had been employed in the source component.

The "asynchronous" USB transfer mode operates completely differently. Rather than making up new data, the data is transferred identically, word-for-word and bit-for-bit. The synchronization problem is solved because the USB link is capable of two-way communication. There is a small data buffer in the USB receiver chip. When the buffer becomes too full, commands are sent telling the media player or PC to slow down the data transfer rate. And when the buffer becomes too empty, commands are sent telling the media player or PC to speed up the data transfer rate.

This is a non-trivial task. It has been done for a few years by "professional" studio equipment via the FireWire link, using special device drivers written for specific operating systems and specific hardware. But for USB audio the published specification not only allows for this data mode, but it is natively supported by all of the major operating systems—Windows, Apple's OS X, and Linux—without the need for any exter-

nally supplied device drivers.

The only two companies to crack this nut to date are Wavelength Audio (www.wavelengthaudio.com) and dCS Ltd. (www.dcsLtd.co.uk). dCS's solution is only found in their products.

Wavelength's solution is not only found in their products but also in their licensees, which at this point include Ayre Acoustics (www.ayre.com) and Aesthetix (www.aesthix.net).

For more information on this topic, please refer to a white paper available on our website at: http://www.ayre.com/pdf/Ayre_USB_DAC_White_Paper.pdf.

Mr. Didden's error mistakenly gives undeserved credit to Benchmark for something they did not create and do not use, and perhaps worse, steals credit from the companies that did develop and perfect this important new system that finally allows for a true low-jitter link between a data storage device and an external DAC.

*Charles Hansen
Ayre Acoustics, Inc.*

Jan Didden responds:

I'd like to respond to Mr. Hansen's lengthy reaction, but will try to keep it somewhat shorter. Let me state, though, that I have no interest in either Benchmark Media Systems or Ayre. Also, my "question" in the interview only mentioned Benchmark as one example of a way to address the jitter problem, only to steer the discussion to that topic. I could have mentioned others, and, in fact, Professor Hawksford mentioned the Linn Klimax (www.linn.co.uk).

Both the Benchmark Media DAC1 series as well as Ayre's QB-9 use an asynchronous process to convert audio samples from digital to analog. Both use a dedicated clean clock at the DAC, and neither uses a PLL (which would introduce jitter of its own). Where the systems differ is that in the Benchmark DAC the samples come across from the PC in a stream determined by the operating system, into an Asynchronous Sample Rate Converter (ASRC), from which interpolated samples are read out as dictated by the clean DAC clock. In the Ayre DAC, samples are coming from the PC as needed to keep a temporary buffer store filled, which is then read out as dictated by the clean DAC clock. My wording of the "question" was incorrect. Instead of "Their USB interface apparently works the same in that it actually requests samples from the media player or PC,

at a rate dictated by the clean DAC clock," it should have read: "Their USB interface apparently achieves the same results in that it actually requests samples from an ASRC device, at a rate dictated by the clean DAC clock."

Significant here is the separation between the arrival timing of the samples and the clean, stable DAC conversion clock. The Benchmark ASRC requires estimating the sample rate of the incoming samples, but does isolate the system from any PC-side sample rate issues. Both systems in theory achieve the same jitter performance (complete jitter suppression), although implementation details may cause differences in practice. There is almost always more than one way to achieve a specific solution.

GOOD TIMES

I have enjoyed your magazine for years, starting with *Speaker Builder* and continuing with your current magazine. Over the years I have enjoyed many articles, such as those by Nelson Pass, Bill Fitzmaurice, Jan Didden, and Gary Galo, just to mention a few. I undertook most of Mr. Galo's mods to the Adcom preamp. I've imagined building the Kleinhorn speaker, but only in my dreams. Recently I have built a B1 preamp and an F5 amplifier based on Mr. Pass' designs. Being a hobbyist, illiterate in electronics, I have enjoyed endless hours of tinkering

because of your magazine. My latest adventure has been to follow along with Kit Ryan's CD Player mods ("Modding the Music Hall CD25.2," April '09, p. 6). I must say I have come to the same profoundly satisfying results. Thanks a million.

*Jan Scranton
scranton@hhmi.org*

Kit Ryan responds:

Thanks for the kind words on the CD player mods. The best part is that you were actually able to make a difference and hear it! In this day of tiny surface-mount devices and high product development costs, it's still comforting to know that we can change things for the better.

Despite the seemingly advanced current state of electronics, I believe we are still very far away from realistic sound reproduction from recorded material. As soon as you hear a live performance begin, you can immediately tell the difference. That is not to say that recorded music is not enjoyable; on the contrary, only that there is still plenty of room for progress and innovation. Is 3-D holographic audio playback in our future, or even possible? I don't know. But people like you will know real improvement when it comes and that will drive both the hobbyists and the industry. Keep up the good work!

aX

CONTRIBUTORS

Neal A. Haight (Audio Aid: Bridge to a Voltage Doubler, p. 5) resides in California and is a frequent contributor to the *aX* letters section.

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Tim Mellow ("A 25W OTL Tube Amplifier," p. 12) received his BSc Hons Degree in Electrical Engineering and Electronics from Dundee University in 1985. For the last nine years he has been working as an electroacoustical specialist for Nokia UK Ltd. His main areas of interest are in audio hardware, microphones, earpieces, and loudspeakers. He is a member of the Institution of Engineering and Technology, the Audio Engineering Society, and the Acoustical Society of America, and has published seven papers on loudspeakers and sound radiation.

Howard Ferstler (Review: Swans M3 Speaker, p. 18) has written reviews for *audioXpress*.

Tom Nousaine (Review: Dayton T1503A Subwoofer, p. 23) is currently a Contributing Technical Editor of *Sound & Vision*, the largest audio/video publication in the world, and holds a similar position with *Professional Audio Review*. In the past 25 years his work has appeared in *Stereo Review*, *Audio*, *Sound & Image*, *Video*, *Car Stereo Review*, *Mobile Entertainment*, *Road Gear*, *Audio/Video International*, *The Audio Critic*, and *Telephony* magazines. Tom operates TN Communications, specializing in loudspeaker measurement, expert listening evaluation and business communications. He is also Chief Operating Officer for Listening Technology, Inc., which conducts expert 3rd party autosound listening evaluations for automotive OEMs and their tier one suppliers. Mr. Nousaine is a past Audio Engineering Society Regional Vice President and past Chairman of the AES Chicago Section. Tom founded the Prairie State Audio Construction Society, the Society for Depreciation Professionals and has been a long time member of the Southeastern Michigan Woofer and Tweeter Marching Society. Previously, he was Director of Capital Recovery for Baby Bell Ameritech and holds Bachelor and MBA degrees from Michigan State University. Tom is a 2nd generation Finnish-American born in Northern Minnesota and enjoys an in-home sauna year round. In addition to evaluating several thousand stereo, home theater, aftermarket and OEM autosound systems, he also enjoys live music of all kinds having attended hundreds of world-class classical and popular musical performances.

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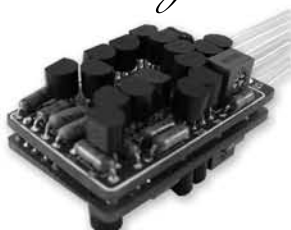
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Ed Simon - AudioXpress, October 2009



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AES Nuggets From New York

Mimicking the overall economic situation, the AES convention rebounded somewhat from the much downsized spring convention earlier this year, but still was smaller than a few years ago. Nevertheless, the mood was upbeat with 321 exhibitors and 18,162 attendees. And the technical program was packed as usual.

DIGITIZING THE GLORIOUS PAST

In this year's convention there were several "red threads," but the ones that caught my attention was a string of several presentations related to the emulation of analog equipment and circuits with digital equivalents. The practice of home recording (and composing), which is almost all digital processing, is definitely here to stay. Yet, there still exists the lure of a good ol' benign distorting tube guitar amp, or the characteristic sound of a 160 ft² EMT reverb plate. With the widespread use of "processing plug-ins" that you can use in many digital workstation applications, the race is on to replace those beloved analog products with a digital plug-in.

This "thread" kicked off with a Master Class by Jonathan Abel and David Berners (Center for Computer Research in Music and Acoustics and Universal Audio, www.stanford.edu), who shared their insights and approaches to analog emulation approaches. Generally there are two approaches. First, you can do input/output modeling. Just measure the I/O relationships of the device you want to model, for instance, by measuring the impulse response at each operating mode or control setting. You can then implement those transfer functions in the digital domain.



PHOTO 1: Lively activities at the Student Section counter.

But this is not as easy as it sounds. Take the example of a simple boost/cut tone control. You can characterize the boost setting at, say, ten positions and extrapolate between them. Then you can do the same for the cut control. You now have 20 impulse responses you need to emulate and extrapolate between. At some point, when each setting is identical, the emulation would give a flat response as boost and cut would cancel.

However, actual measurements of one equipment sample show that this is not the case. There can be significant interaction between controls so that identical settings do not produce a flat response. In our example, to catch the interaction you would need, say, ten impulse responses from each control at each of ten settings of the other control, so now you end up with a requirement for 100 responses. With four interacting controls the number jumps to 10,000 impulse responses that you need to measure, turn into algorithms, and

interpolate between. This doesn't seem like the simple and easy way to do this.

Instead, try a second approach called the Physical Modeling (PM) method. You develop an algorithm that actually describes the circuitry in the component you want to model, and that then automatically includes the interaction. Think of it as a kind of real-time, live, online SPICE simulation of the circuit. Of course, it has its own set of issues. For one, nonlinear and time-varying elements need to be processed separately from linear non-time varying elements. But even the emulation of a simple RC low-pass filter element is not without problems.

In a real system you can, for instance, vary the value of R with a pot and thus produce a variable high cut. But in a digital system, that apparently causes problems with initial conditions and the discrete "jumps" in circuit signal values due to the discrete nature of the Fourier transform. So, it can be done, and some examples were given that

even included the emulation of hum and noise that the real equipment is known to have, but it is certainly not an undertaking for the faint of heart.

EMULATING THE LESLIE

Jorge Herrera et al. from Stanford tackled this one. A Leslie speaker (**Fig. 1**) consists of a woofer that radiates onto a rotating baffle (bottom), while the mid-range radiates into a rotating horn. The rotational speed can be varied. Slow speeds produce chorale type of sound variations, while faster speeds produce a tremolo effect. They did not use the virtual instrument emulation method described previously, but

constructed a large table of impulse responses from the system to a pair of microphones over very many horn and baffle angles.

They changed the baffle and horn positions in small increments, recorded the mike signal, and then calculated the baffle and horn positions afterwards based on the signal arrival times at the microphones. They even modeled the mechanics of the rotation motors (acceleration and deceleration) to produce a complete emulation that reacted and sounded like the real thing. To implement the Leslie effect, the input signal is filtered by time-varying FIR filters derived from the table according to the

time evolution of the horn and baffle rotation angles. It is the state of the speaker rotation controls on the emulated front panel (the power and chorale/tremolo switches) that determines the target rotation rates for the horn and baffle that drive models of their respective rotation dynamics to produce horn and baffle rotation angles over time.

They then interpolated the horn and baffle filter impulse responses from tabulated responses according to their rotation angles. This demonstration provided a very convincing example of a Leslie 44W.

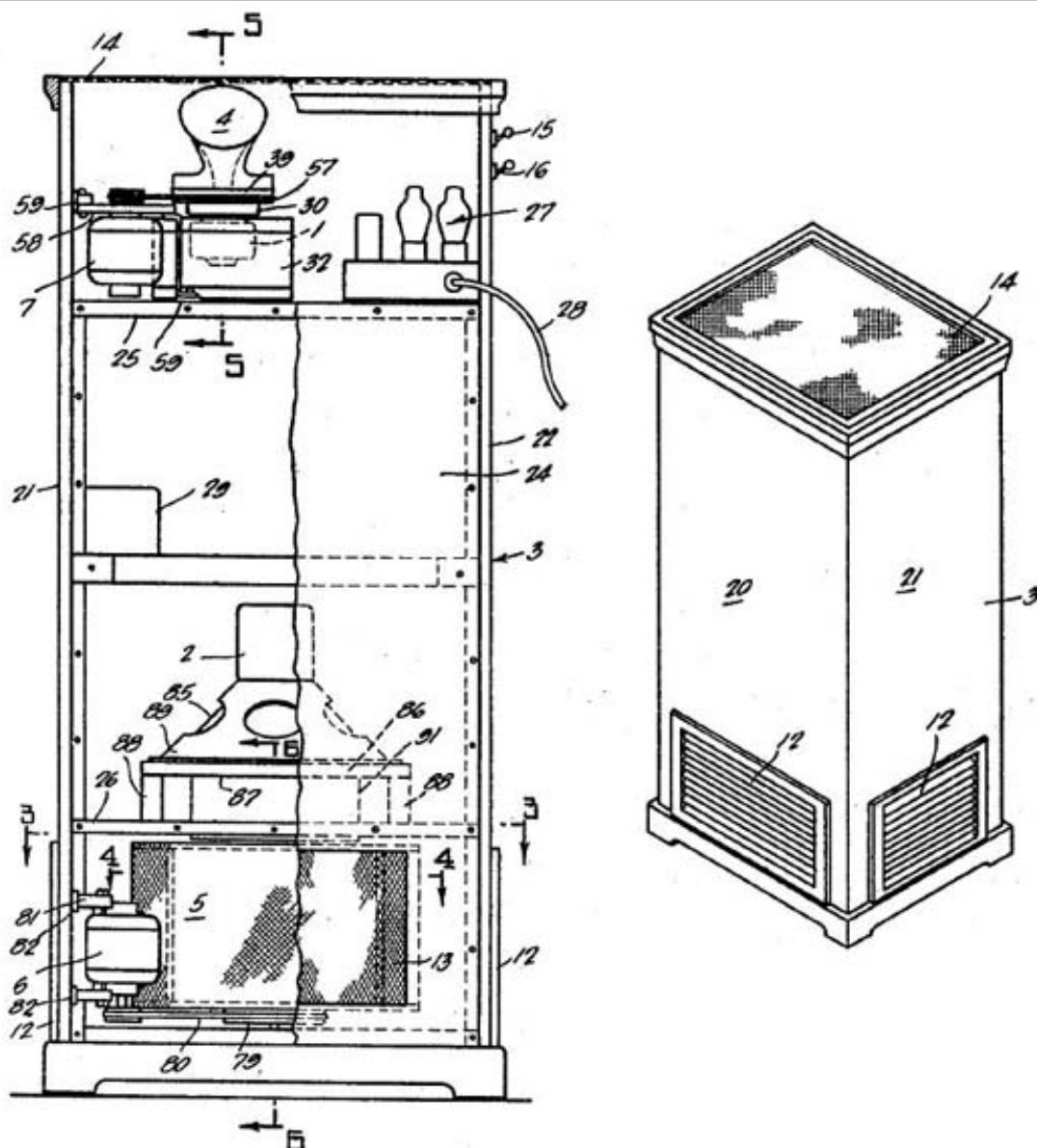


FIGURE 1: The Leslie principle from the original patent.

EMT 140 REVERBERATES AGAIN

The EMT 140 is approximately 6m² (65ft²) of steel plate, driven by a voice coil with another voice coil picking off the signal due to the plate vibrations. There is also a separate, parallel, damping plate. The (adjustable) damping plate controls the acoustic impedance seen by the signal plate and thus the signal decay. The reverberation effect is determined from the signal decay rate when it travels across the plate as well as from the latency, caused by the time it takes the signal to travel from actuator to pick-up.

This is a system that, with some approximation, you can consider to be linear and time-invariant, which makes emulation so much easier. Again, you must decide whether to emulate this system by measuring its impulse response for many different damping plate positions, or use Physical Modeling as discussed before. The first method would then include filtering the signal through the impulse response, interpolating between tabulated impulse responses for the various damping plate positions. Because of the very large decay time of up to eight seconds, this is quite costly in terms of memory and computation requirements.

Physical Modeling would entail solving a system of partial differential equations that describe the plate dynamics. This would allow modeling the nonlinear regions of the plate, but is also computationally costly. The authors finally chose a hybrid system based on earlier work and limiting the model to the plate's linear operating region. You can find the detailed system design in Conference paper 7828.

GUITAR AMP TUBE ROLLING—AT THE PC

In their quest to faithfully model guitar tube amplifier stages, Cohen and H  lie from IRCAM in Paris (www.ircam.fr) and Orosys in Montpellier (www.oroys.fr) went deep into the details. When they used standard Leach or Koren tube models, there always remained an audible difference in character between the digital simulation plug-in and the real thing. This is mainly due to some nonlinearities that are responsible for

the “sound signature” of the original equipment, which results from the designers’ choices, such as location of the tone controls, specific cutoff frequencies, biasing of tube stages, and so on. In their study, Messieurs Cohen and H  lie concentrated on the behavior of a typical 12AX7 triode stage of a high gain preamplifier. Including the effects of internal capacitances and in particular the Miller capacitive feedback effect brought the simulated sound much closer to the original sound.

In the end, it proved necessary to even include the effects of grid current in the model to get close enough to the original. Interestingly, the results from running their detailed model through Linear Technology’s (free) LTspice gave the same results as that produced by their digital VST plug-in, confirming the soundness of the approach. Conference paper 7929 has the full details, while you can listen to sound samples based on either model, with or without the Cgp or Ig modeling, at www.orosys.fr/cohen/aes127.htm. They promised to include transformers in the next iteration of the modeling effort.

THE TUBOID PLUG-IN

Despite the paper’s unlikely title, “Over-threshold Power Function Feedback Distortion Synthesis,” this was another packed presentation on digitally modeling analog equipment (Tom Rutt, Coast Enterprises LLC, NJ at www.coastin.com). Or, more accurately, emulating the nonlinear distortion of a typical high-gain tube amplifier stage. (As an aside, I find it interesting that after the initial euphoria—decades ago—about how digital processing would allow almost distortionless sound, this same digital capability is now used to faithfully re-create the distortions we once despised). This was another case in which the importance of the effects of grid current in tube amplifiers to obtain faithful emulation was noted.

In a tube stage, grid current would cause a lowering of the effective input signal due to signal losses across the source resistance. The effect is like an increase in feedback when grid current starts to flow. Tom has a patent on a

hardware realization which has been abandoned. He now modeled it as a VST plug-in. The system contains a high-gain tube stage with a linear feedback network.

However, when the output signal of the model surpasses a positive or negative threshold (when grid current would start to appear, modeling the grid current flow. **Figure 2** shows how a relatively complex circuit is required to faithfully emulate such a simple phenomenon as grid current! Details are in Conference paper 7930, while you can hear sample sound clips that demonstrate the effect at <http://www.coastenterprises.com/OTPF/demoClips>.

THE POWER OF SUGGESTION—AGAIN!

I attended a workshop on “Audio Myths” in my ongoing quest to find the parameters that determine audio quality as perceived by the listener, and to resolve the paradox that masking makes it difficult, if not impossible, to hear sounds 40 or 50dB below the loudest level. Yet many people claiming to hear artifacts 100dB down.

Poppy Crum (Johns Hopkins School of Medicine) played the famous Beatles recording some people claim has a hidden message that can be heard when played backwards. She did play it backwards, but for most of us (except those who knew the supposed message) the reverse sound was completely unintelligible. Next, she played it again, but now with the supposed text shown, Karaoke-style. The words were clearly heard, no mistake possible, it’s very easy to hear if you know what to expect. Listening really takes place between the ears rather than in the ears.

DROPPING THE GAUNTLET

Siegfried Linkwitz took a different approach to achieve better stereo placement and imaging using a traditional pair of speakers. Mr. Linkwitz has done a lot of research into the optimum radiation pattern and placement of loudspeakers for home stereo reproduction. Linkwitz Lab’s products from that research, notably the Orion dipole speakers, have earned a strong reputation for excellent stereo reproduction (www.linkwitzlab.com).

In his paper (Conference paper 7959) Siegfried recapitulates what is known to determine stereo imaging, placement, and reproduction of venue acoustics, as well as the “white spots” in our understanding of these issues. He issued the following challenge to the audio community: characterize the differences in phantom image creation and loudspeaker/room masking for representative speaker radiation patterns and room placements; determine the sensitivity of the results to loudspeaker placement closer to, or further away, from the walls; explain the results in psychoacoustic

terms; and suggest improvements in the radiation pattern, implement them and verify their effectiveness.

Of course, all this would need objective listening tests, involving large groups of listeners. The collected data would need to be evaluated statistically to resolve differences between previous studies and to give conclusive answers to questions about radiation patterns—not a simple undertaking for a rainy Sunday afternoon, but nevertheless worthwhile as another step forward to our understanding of these issues.

BLIND CONFIDENCE

Jon Boley and Michael Lester (LSB Audio, www.lsbaudio.com, and Shure, Inc., www.shure.com) presented a different way to interpret results from an ABX-double-blind audio test. Recalling that the first ABX test was documented in 1952 by the Naval Medical Institute (and you thought ABX was a Hi-Fi audio development!), they addressed the eternal problem of how confident you could be that a certain outcome proves your hypotheses. There is never absolute certainty.

If, for instance, statistical analysis says that the confidence level is 95% that there was an audible difference, there is still a 5% chance that the result is based on pure guesswork. You can try to get your confidence level up to, say, 99%, but that would require a massive test with large numbers of trials and test subjects, rapidly making the test unwieldy. Boley and Lester looked at the test results with a statistical method based on Signal Detection Theory (SDT). Their point was that with SDT you can get a higher confidence level from the same number of test results than with the usual statistical analysis. Details, with lots of math, are in Conference Paper 7826.

ANALOG IS EVERYWHERE™—AND SO IS JITTER!

I hope that the powers at ADI will forgive me this little frivolity, but it is true! We all know about digital audio’s fight with jitter in reconstituting digital audio samples to a nice analog waveform. But what is jitter, other than time modulation of an analog signal? Malcolm Hawksford has made the case that, for example, slew rate limiting in a DAC current-to-voltage converter looks and sounds like jitter¹, and this is from an analog circuit.

Richard Tollerton from Isomorphic Software, Inc decided to analyze tracking distortion resulting from the misalignment of a phonograph playback cartridge. He developed a highly accurate simulation of both the vertical and the horizontal tracking distortion, which allowed him to evaluate tracking distortion with real music content. It turned out to be equivalent to digital audio sampling jitter! The author

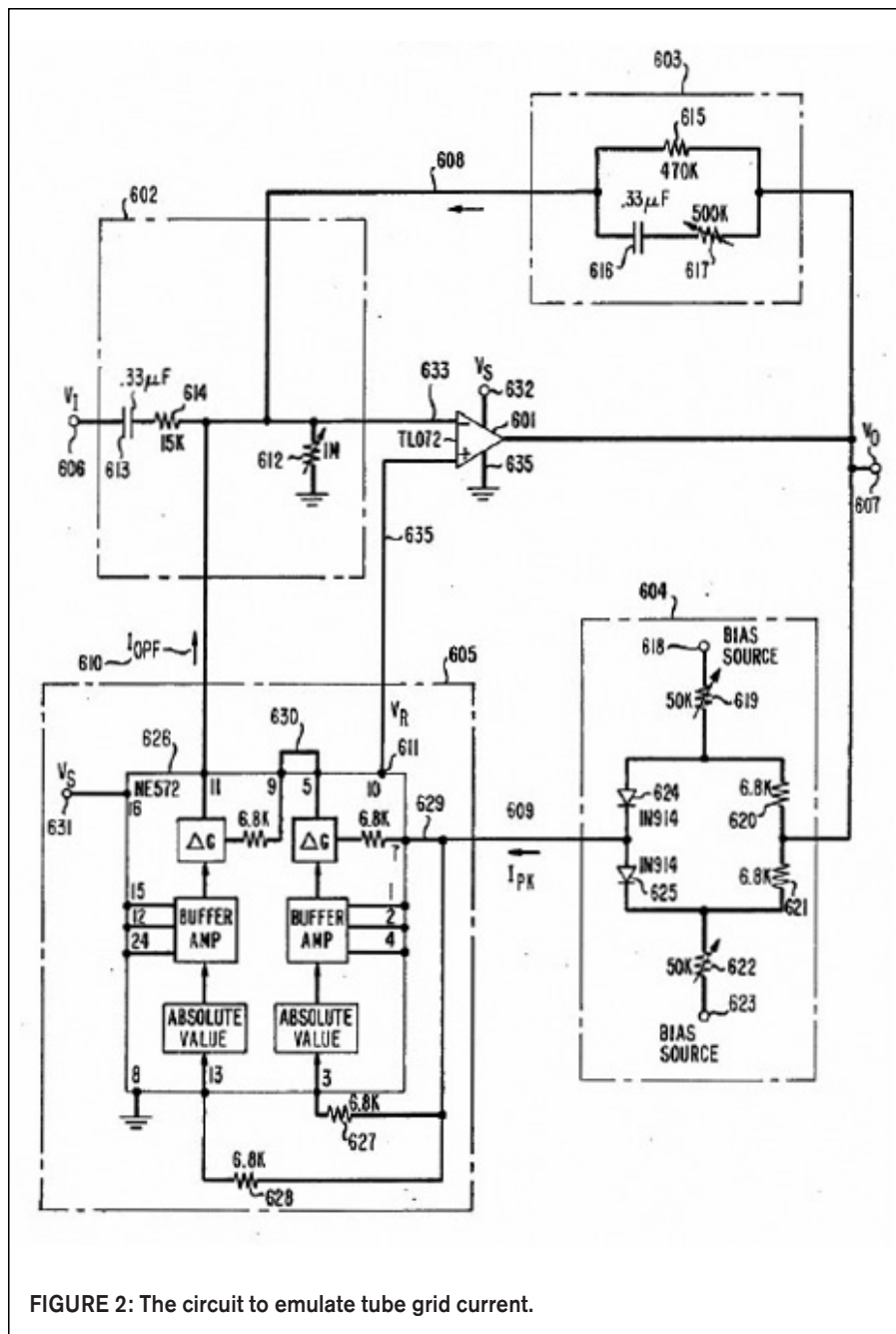


FIGURE 2: The circuit to emulate tube grid current.

described some anecdotal listening tests that appear to confirm audibility of real-world tracking errors, but this clearly needs (much) more work to be conclusive. The jitter spectrum, by the way, is the same as the signal spectrum which Malcolm Hawksford believes is one of the more objectionable jitter spectra.

If you want to maintain that LP playback sounds better than digital playback because it doesn't contain jitter, you better think again! Conference paper 7924.

THAT'S INTERESTING!

People involved in solid-state (multi-channel) analog volume controls are constantly on the search for better/cheaper/easier-to-use solutions. IC on-chip switch and resistor technology advances have all but replaced analog VCAs based on two- or four-quadrant analog multipliers with what for all intents and purposes are stepped attenuators on a chip. The issues are the same as in a discrete switch implementation: accuracy of the resistor ratios in the ladder, especially with very large resistor ratios if you want more than, say, 60dB of range; nonlinearity of the switches and resistor elements, variation of the load impedance presented to the source, and the best way to mate the attenuator network with a line or buffer amp. (In some cases, IC designers find ingenious ways to linearize switches and resistors, as you can read here: <http://www.jeffrowland.com/Technology/DigitalVolumeControls.htm>). Most application designers would like the freedom of being able to mate their own favorite integrated or discrete amp to the switched attenuator network, but the chip designer likes everything on his chip so he can tweak the total performance.

Several years ago Wolfson (www.wolfsonmicro.com) marketed a switch-network only chip, but that was discontinued after some time. Currently, among others, the PGA23xx product line from TI (www.ti.com) and the CS3308/3318 from Cirrus Logic (www.cirrus.com) offer integrated solutions that truly have spectacular performance with > 120dB dynamic range,

PHOTO 2: AES President-Elect Diemer de Vries and 127th Convention Chair Agnieszka Roginska.



gain/attenuation ranges from >20dB to <-100dB, and vanishingly small nonlinearities. They also come in packs of up to eight channels with excellent tracking between channels and an easy serial interface to a microcontroller. These chips are well suited for multi-channel theater/home sound systems and for recording and mixing consoles alike.

Now THAT Corporation has come out with a single-channel balanced attenuator, the THAT 1571, which is primarily meant to be mated with their low-noise differential audio preamplifier THAT 1570 (www.thatcorp.com). What makes this part unique is the ability to digitally set the gain over 56 1dB steps from 13.6dB to 68.6dB, which is much higher than other solutions I am aware of and is required for mike preamps. It is also fully balanced, and can be mated to other amp blocks if needed. I managed to score a sample so will definitely put it through its paces. (By the way, the larger chipmakers such as ADI, TI, and National were notably absent from the convention. Another sign of the times?).

25 YEARS PRECISELY

Audio Precision (<http://ap.com/>) was celebrating its 25th birthday, not only with champagne and cake but also new product releases. One that caught my eye was an ultra-wide bandwidth option for their APx line. Option BW52 extends the FFT bandwidth to 1MHz. At 24 bit, that is. This will allow designers of class-D amps, D/A converters, and switching power supplies not only to look at the switching frequency spectrum, but at several harmonics as well.

As I mentioned, attendance at the AES seems on the rebound, and that is a good thing. Exchanging information and experiences with your peers, combined with the chance to directly interface with many manufacturers, remains an important *force multiplier*, if I may borrow this military term, for those in the industry. It remains, literally and figuratively, a matter of making the right connections.

REFERENCE

1. Interview with MJH Hawksford by the author, published in *audioXpress*, Nov. and Dec. 2009 issues, and *Multi Media Manufacturer* 5/09 and 6/09. *aX*

A Low-Cost, High-End Music Server

Computer-based audio is making inroads into the mainstream audio market. Once considered the domain of compressed MP3 files of “pop” music, there are several high-end music servers that can store and manage sizable collections of music files. These devices carry the usual hi-end price tags, costing—at a minimum—around \$7000. They are big on organizing files on several different characteristics and having user-friendly features, such as touchscreens, but they work only with MP3 files or current “red book” files. They’re great for storing and retrieving what is currently available to consumers, but, for the money, don’t offer a path to the future (i.e., the ability to store and play back the new hi-res files available online).

OPTIONS

All this prompted me to check into the cost of putting together a computer with the capability to download and play these hi-res files, especially the “HR X” files offered by “Reference Recordings.” I found that for considerably less money (about \$3000), I could build a machine to store my current LPs, my CDs, and new files that I would download. The only additional piece I needed was an external DAC to connect my stored files with my stereo.

I thought of building my own computer, making this project like the kits I built from Dynaco, Heath, and Eico that got me started in this hobby years

ago. But a quick trip to the computer store eliminated that option. It would cost almost \$2000 to build one versus \$800 for a Dell XPS. Actually, I went one step cheaper.

I had a Dell Dimension sitting at home with a burned-up hard drive. While the CPU wasn’t the latest, it was still fast enough to handle streaming music. A replacement drive with 500GB cost me \$130, with an external backup drive costing another \$150. I was well on my way to my low-cost music server.

To complete the computer setup, I needed a sound card that could handle sampling rates up to 192kHz and bit depths up to 24. The highest sampling rate I could find on a sound card at the time was 94kHz at 24 bits. It also required the use of a USB interface, which is currently limited to 94kHz. Taking a hint from the Reference Recording website (www.referencerecordings.com), I looked into Lynx products (www.lynxstudio.com). They make equipment for recording studios, and seem to be popular with hobbyists, too.

The Lynx card that came closest to what I wanted was an AES16. It was a bit of overkill because it had a built-in mixer, the ability to use an internal clock—or that from an input data file—and could handle eight stereo channels. The list price (around \$800 with cables) testified to its “professional” status. With several of these cards for sale on eBay, I was able to get one for almost half price.

SETTING UP THE PC

Installation is simple. The Lynx card fits in a normal PCI slot on the main board, just like any other sound card. The external face of the AES16 has two D connectors marked A and B. There is a matching D connector on both of the supplied cables, each of which has four input channels terminated in female XLR connectors, four output cables terminated in male XLR connectors, and one BNC connector carrying the clock I/O signal. You’ll need only one cable for a stereo hookup.

You can download the necessary software from the website, as well as any firmware updates and an application to verify that the card is working correctly. The installation software will leave an icon on the desktop called “L mixer.” Double-clicking this will bring up a screen with three panels titled “Output Screen,” “Mixer Adapter,” and “Record/Play.” On the record/play panel, clicking on DI1L will bring up a list of available inputs. You should select digital in 1L. Similarly, you should select digital in 1R for the DI1R button. After setting the depth to “auto,” you can mute the other seven channels.

The output panel is easier. All that’s necessary is to mute everything except 1L and 1R. In the mixer adapter panel you should select “internal clock” for the preferred clock source. This reclocks the signal in order to reduce “jitter.” You also must select the rate on this panel. This is the sampling rate of the file you want to listen to.

In order for all this hardware to function correctly, you need some software to maintain and organize the music files. One of the popular (and free) programs is called MediaMonkey, which is the one I used. The free version is adequate for most uses and is easy to install. Just go to their site (www.MediaMonkey.com) and download the program.

To get things rolling, you must tell "MediaMonkey" where you keep your music. Simply start the application, click on file menu, and select "add/rescan tracks to library." You will then have the opportunity to tell the program which PC drives to scan and which file types to look for. If you use the free version, you'll need to do this each time you add a file to the PC. The other version does this automatically each time you start the program.

Selecting the proper output takes a few clicks more than loading the library. In the tools menu pick "options," then "player," then "output plug-ins." From this point you should double-click "MediaMonkey direct sound output" to obtain a list of the devices available. If the sound card is set up correctly, you should see eight Lynx devices in the list. Choose the one you are going to use (device 1 if you set it up as I did). While you're on this page, make sure the "remove silence at beginning/end of track" box is not checked. If it is, you will not be able to mix different resolution files in the same playlist. It seems that MediaMonkey doesn't check the file header when this option is on or doesn't tell the DAC to change the resolution.

CHOOSING A DAC

The choice of the AES16 card pretty much dictated my choice of a DAC. Not only must the DAC handle multiple sampling rates and bit depths, but it also needed to have AES/EBU inputs to accept the signal from the PC and regular RCA outputs to mate with my stereo preamplifier. This choice of sound card also eliminated several good-sounding DACs, such as the PSAudio Digital Link III (www.psaudio.com), from consideration. Cost also eliminated others such as the Berkeley Audio Design Alpha DAC (www.berkeleyaudiodesign.com) and the new

Perfectwave DAC from PSAudio.

I settled in to listen to the DACs that fit my criteria because it is important that the DAC I chose have as good sound as my ancient Audio Research DAC 1/20. I've listened to other DACs and CD players over the years, thinking that I should update, but none of the newer equipment sounded any better than what I already had. Because the new DAC would be responsible for the playback of my existing CDs as well as the new hi-res files, I didn't want to take a step backward in sound quality.

The Briston external BDA-1 (www.briston.com) fit my needs exactly. It had all the input types anyone would need: SPDIF (RCA and BNC connectors), AES/EBU, optical, and USB. It had balanced and single-ended outputs, and would lock onto any commonly used bit depth/sampling frequency combination between 16/44.1 and 24/196. The fact that the USB input could not handle more than 16 bits was not a problem because I wasn't going to use it anyway.

I've owned Briston equipment before and thought it sounded fine. I wasn't disappointed with the sound of the DAC either. It was a little more "etched" than my ARC, but not disturbingly so. My quick test for listenability is to play some Cecilia Bartoli recordings. If the richness of her voice disappears and she begins to sound like a witch, the DAC or player is not for me. The Briston was just fine on this, and subsequent, tests. This choice completed the physical system.

THE REST OF THE SYSTEM

Preamp: Audio Research (www.audioresearch.com) SP-8 MK-III

Amp: Audio Research D90-B

Crossover: Audio Research EC-3 (6dB/8VA at 100Hz) or Marchand (www.marchandelec.com) XM26 (24dB/8VA at 70Hz)

Speaker: Quad -63 ESL (www.quad-hifi.co.uk)

SW: Homebrew (2 × 12" Audio Concepts (www.audioconcepts.com) dual voice coil in a 15 ft³ cab)

SW Amp: Crown D150 (bridged for mono) (www.crownaudio.com)

CD Transport: PS Audio Lambda 2

I'm not fussy about interconnects or speaker wire, which I usually pick up at Radio Shack (www.RadioShack.com) or similar retailers.

The speakers sit with their centerlines about 6' apart on Arcci stands. They are angled to fire directly at the listener, about 6' away and on the centerline between the speakers. To accommodate the low vertical height of the center of the speaker, I needed to construct a futon so that my ears would be the correct height. The speakers are also baffled on the outer edges by 18" wide by 6' tall panels to help reduce low-frequency cancellation of the dipoles.

While I started my listening with the ARC crossover, I ended up using the Marchand exclusively. I liked the openness of the ARC, but the available crossover points (50Hz or 100Hz) and slope were not optimum for these speakers.

LISTENING TESTS

I was unable to perform level-matched, double-blind tests. I don't have an ABX box, nor do I have access to a group of people willing to sit and listen to repeats of the same music, looking for differences. I did as well as I could matching timing and level between the AES/EBU input (PC) on the DAC and the SPDIF input (CD transport).

With everything assembled, I settled back to listen. Before the music even started, I was impressed. The fan on the computer was inaudible from the listening position. This had been an early concern, which, fortunately, I didn't need to worry about. I started off with one of the Reference Recordings HR-X DVDs. I put on Crown Imperial (HR-112) and immediately heard the effect of the higher number of bits.

It seemed as though I'd arrived late to the performance and had to listen to the first number from the lobby. I needed to increase the gain considerably to get to the usual listening level (around 80-85dB at the listening position). The sound was rich, full, and easy to listen to. It also seemed to come from a "blacker" background. The timbre of the instruments was

much more genuine in the HR-X version.

For comparison sake, I also purchased the standard CD of this recording. The sound was less smooth than the HR-X recording, and had more of what people refer to as CD "glare." I would not have noticed this if I didn't have the "hi-res" recording for comparison.

I ran similar comparisons on other HR-X recordings (HR-117 Percy Grainger, HR-116 Thinking of Bix, and HR-96 Rachmaninoff) with similar results. The large-scale works appeared to be recorded from well back in the hall, but the recorded image of the piano on HR-116 was right in my lap. I've never heard a better recording of a piano. The other Reference Recordings with Dick Hymen playing solo piano (RR-33DCD Dick Hymen plays Fats Waller and RR-50DCD Dick Hymen plays Duke Ellington) have long been favorites of mine for their great piano sound, but they take a back seat to this recording with its more life-like sound.

I also tried downloading what I call the "mid-res" recordings (96/24) of the same recordings, but found it difficult to make comparisons with the "hi-res" files due to shortcomings in my setup. The AES-16 sound card would not pick up the bit depth or sampling rate from the selection being played, because it was not being input through what the card calls its input, so I needed to use an internally generated sampling rate. This can be set only when the music control program (MediaMonkey) is stopped, making it impossible to set up a playlist with recordings with different resolutions. The differences between the "mid-res" files and the CDs were less noticeable, but still there.

CONCLUSION

The WAF (wife acceptance factor) is low. I have a dedicated listening room, so this has a low priority for me. For those who have their stereo in the living room—perhaps integrated with the TV—this type of setup would probably not work. If you really want to enjoy

the sound afforded by the hi-res files, you might be better off spending the extra money for a more user-friendly solution.

The not-so-good was partly my fault. When I purchased the AES-16 on eBay, I didn't have a DAC with XLR connectors, so I couldn't check the playback. The recording part was fine, but when I got the Bryston BDA-1, I found that the playback didn't work and it was too late to do anything about it via eBay (see Bruce Brown's article in the Aug. 2008 *audioXpress* about buying equipment on the web.)

There is a "silver" lining to this, however, because I was pleasantly surprised by the tech support at Lynx. The tech logged onto my computer over the net (I forget what software he was using), quickly diagnosed it as a hardware problem and gave me an RMA. In less than two weeks I had my card back with a new IC on it. It cost me \$100, but it was worth it, if only to find that there is good product support out there.

There are a few improvements I'd like to make that really don't affect the playback but would make things more comfortable. I'd like to get a wireless hub so that all the computers in the house can share the Internet. I'd also like to get a control program that I can put on my laptop so I can sit in the listening seat and set up and control the playing from there. PC Anywhere from Microsoft is long gone, but there are free programs out there that will accomplish the same thing.

On the plus side are: the sound and the cost. For less than \$3000 I was able to assemble a system to play hi-res files and store all my digital data. I didn't need to buy a PC, which would have added only \$800 or so to the total cost. To me the pleasure of listening to these files is worth that cost.

I can't express how much I enjoy the improvement in the sound of my system except, perhaps, by analogy. I feel like those audio enthusiasts of the 1950s who were among the first to hear stereo via their two-track tapes, except that I've done them one better. There is no degradation of my copy and there never will be.

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