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BIRTH OF A TUBE

Blackburn Unveils the E813CC

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New Approach to Tube Design

The famous Blackburn MicroTech Solutions tube factory has launched the first in a line of audio tubes featuring the first radical re-design in tube technology in over 40 years.

When I first heard that tube production was starting up again at one of our great British institutions—the famous Mullard tube factory in Blackburn—I was rather excited, to say the least! To my ears the sound produced by tube equipment is simply unbeatable.

One of the reasons, I believe, that tubes sound so good is that, simply put, the electrons have plenty of space to move around—i.e., a low current density. On the other hand, solid-state devices squeeze all those electrons across tiny p-n junctions and the like. It's a bit like egg production where transistors are the battery hens, while tubes are definitely free-range! However, before all you semiconductor aficionados start reaching for pen and paper, I must stress that solid-state devices do have an equally important place in hi-fi—an all-tube

no-transistor DAC could certainly generate a lot of heat!

A BIT OF HISTORY

The story begins in 1919 with the collapse of the Z Electric Lamp Company of Southfields run by Captain Stanley R. Mullard! The following year, Captain Mullard set up the Mullard Radio Valve Company Limited.

In 1924, still wanting to expand as the demand for tubes continued to grow, Captain Mullard sold half his shares in the Mullard Radio Valve Company to NV Philips Gloeilampenfabrieken of Eindhoven, Netherlands. This injection of capital allowed for the construction in Blackburn of what was to become the largest tube assembly works in the world. When the doors opened in 1938, the new Mullard Radio Valve Company Limited employed 28 staff. This grew

to about 3,000 people by 1945, and by 1954 the site was capable of manufacturing 500,000 tubes per day.

At its peak in 1961, the Mullard site generated its own gas and electricity, making it fully independent of the local town supplies. Over 6,200 people were employed both in Blackburn and the various feeder factories supplying raw materials and sub-components to the tube works. Sadly, due to the demise of the tube in favor of the transistor, by 1984 all these factories had closed or moved to other activities. The Mullard organization changed its name to Philips Electronics Components (the main shareholder) and continued to make and develop cathodes for use in cathode ray tube (CRT) applications.

Alas, even the good old CRT is now history for domestic televisions since solid-state LCD and plasma displays



PHOTO 1: Close-up of the new E813CC tube.

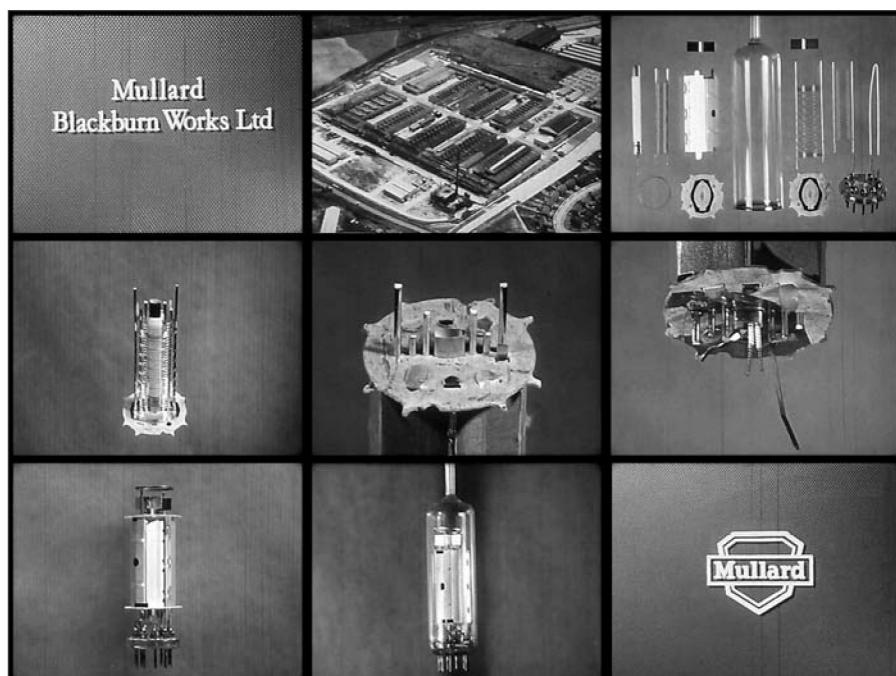


PHOTO 2: The Blackburn story collage.

have taken over in recent years as display devices. This might have marked the end of an era, had it not been for those astute people at Blackburn who realized that there is now a growing demand for tubes to be used with high-end audio equipment. The company's name was changed once again, this time to Blackburn MicroTech Solutions. With all the local expertise (indeed some of the engineers served apprenticeships in the original workshops!) and background knowledge of tube manufacturing and cathode development, the Blackburn factory turned its attention to audio tubes with the ambition of once again being at the leading edge of tube development.

THE DAWN OF A NEW ERA

The company is, of course, no longer able to use the "Mullard" name because this brand is still owned by Philips Electronics in the Netherlands. However, with the launch of the TechTube brand, Blackburn MicroTech Solutions hopes that the heritage and tradition associated with the Blackburn factory will be revived. In fact, 2008 saw the launch

of the first real tube development since its initial launch back in the 1920s, with the production of the E813CC double triode.

The TechTube E813CC is functionally identical to the ECC83, but the new designation gives a clue that this is not simply a "me too" copy of an old favorite.



PHOTO 3: The naked tube!

What those clever chaps at Blackburn MicroTech Solutions have done is to channel all their expertise into electron gun technology, resulting in a radical re-think of the design of a triode tube.

Blackburn MicroTech Solutions has put a copy of an original Mullard history film on their website showing how



PHOTO 4: Close-up of cathode assembly.

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their tubes were made. By examining frame captures from the film, you can see that a standard indirectly heated triode audio tube is constructed as a set of concentric cylinders. In the center, there is a filament that is inserted into a thin metal tube to form the cathode, which is electrically insulated from the filament and has a special coating to facilitate the

release of electrons when heated by the filament.

Around the cathode is the grid, which is a larger diameter tube made of fine mesh or a spiral of wire that the electrons pass through to reach the plate. The plate is an outer tube of metal that is the final destination of the thermionic electrons as they pass through the tube

elements. This traditional electrode assembly is supported by two insulating mica washers—one at the top and one at the bottom of the cylinder. By making the plate positive with respect to the cathode, the electrons are attracted to it and their flow is controlled by a small potential applied to the grid—simple!

However, Blackburn MicroTech So-

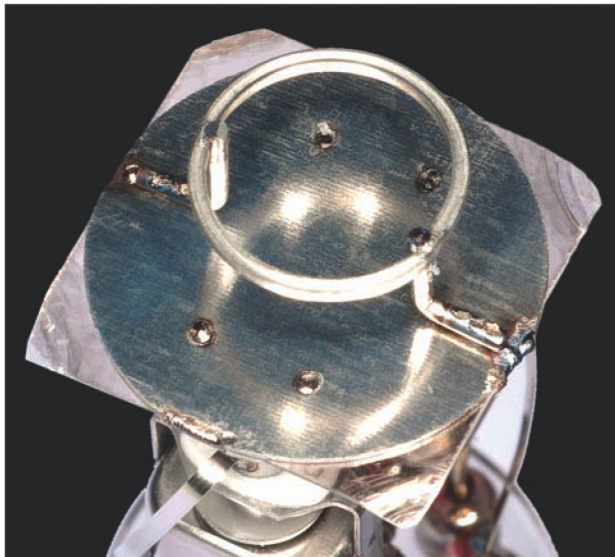


PHOTO 5: Close-up of getter and mica upper support.

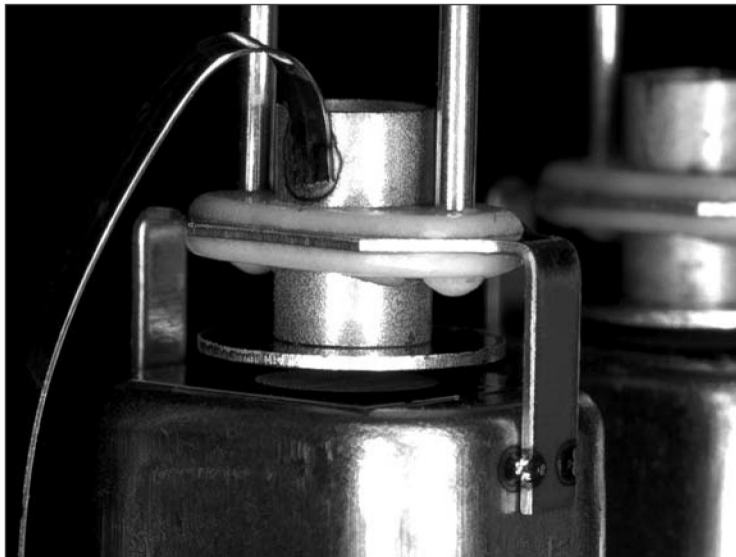


PHOTO 6: Close-up of grid assembly.

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lutions has gone back to the drawing board and developed their own structure for triode assemblies. It is a completely new way of making triodes based on microassembly techniques previously used in the CRT industry. The result, they claim, is more accuracy and, most important, more consistency.

A NEW TUBE IS BORN

The tubes come in a well-designed cardboard box that elegantly does the job of holding the tube securely without going "over the top." I am very happy that not too much of the cost of the tube is being diverted into fancy packaging. Having said that, the blue and white printing on the tube is very refined, and the use of the Union Jack on the box leaves no doubt as to their origin!

From the close-up photographs of the electrode assembly of the new tube, you can easily see how different it looks from a conventional tube. The filament and cathode assembly, together with the grid, are located inside a metal can. Instead of being emitted radially as in a conventional tube, the electrons are di-

rected through an aperture at the top of the can that has the grid laser-welded to it. The plate that collects the electrons is situated on the other side of the grid just above the can and is in the form of a little metal "top hat."

So the elements of the tube are of a planar construction, instead of the conventional radial design, which is strikingly similar to the design of electron guns used in CRTs. This is, perhaps, not surprising when you consider that the factory has been specializing in electron gun components for the past 40 years!

This planar tube design results in a tube with very low capacitance and minimal structural restraint. The low capacitance bodes well for a tube with excellent high-frequency characteristics. The fact that there is only one mica support at the top of the tube, instead of the more usual two, should reduce the mechanical coupling of the electrode assembly with the outer glass envelope and, therefore, reduce coloration through acoustic feedback to the tube. However, you must balance this against the risk of microphony caused by a reliance on



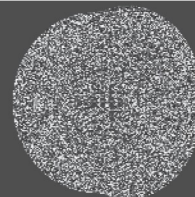
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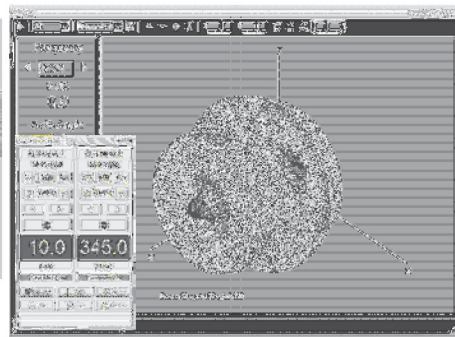
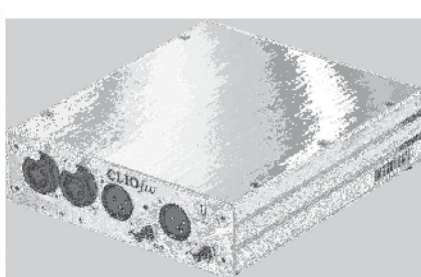
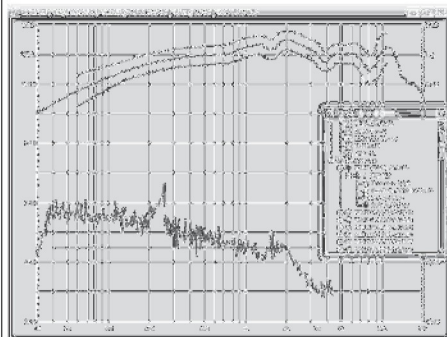
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the internal structures for support of the electrodes. The term "microphony" describes the injection of a metallic ringing into the audio signal and is caused by mechanically induced vibrations being transmitted from outside the tube to the electrode assembly inside.

Although the E813CC has the same electrical characteristics as the 12AX7/ECC83 and also the identical B9A pin base and 6.3V heater for each triode, it is constructed to much tighter tolerances. Blackburn MicroTech Solutions claims that the tubes will have extended life. Also, the variability in performance between tubes has been dramatically reduced. Due to the small surface area of the tube and the micron cathode and plate-to-grid distances, the internal capacitance is very low indeed, with a quoted grid-to-cathode capacitance of 0.4pF (25% of that of a traditional tube) and a grid-to-plate value of 0.9pF (50% of that of a standard tube).

Finally, the filament of each triode consumes a mere 112mA. This is about 75% of the heater current of a standard tube, so the tubes should generate a bit less heat, which is a good feature. Of course, "different" doesn't necessarily mean "better," so how exactly do the TechTubes stand up against the competition?

THE TECHTUBE ON TRIAL

After some extensive listening with my top-of-the-range conventional audioophile tubes (with a price tag to match!), I decided it was time to swap the three 12AX7s in my phono preamplifier unit with the new E813CC tubes. Straight out of the box, they certainly

sounded impressive in the lower registers. The bass performance was crisp, tight, punchy, and well extended. The bass drum in my well-used copy of Stravinsky's "The Firebird Suite" (The Atlanta Symphony Orchestra—Telarc digital recording DG-10039) was truly breathtaking!

Initially, the top end was slightly muffled and the midrange was not as clear as I had been used to. Vocals were recessed in the sound stage and there was a general feeling of restraint to performances. I was expecting quite the reverse, given the low capacitance construction.

However, after a few hours of running in, the situation changed dramatically. The bass performance was unchanged, but the top-end reappeared from nowhere.

Following a full 24 hours of running, the TechTubes had clearly bedded in well. The muffling had gone and the tubes produced a sound of effortless clarity and openness. Vocal soloists took up their rightful place in front of the orchestras again and electric guitars demonstrated fantastic speed and detail in all the harmonics. This is not to say that they are at all harsh.

In fact, Blackburn MicroTech Solutions seems to have achieved the wonderful tube smoothness without sacrificing clarity and detail. On jazz recordings, "hi-hats" sound crisp, fast, and sparkling. The bass notes in rhythm sections are clearly defined and well proportioned with the rest of the music.

A direct-to-disc live recording from the 1970s of "Lincoln Mayorga and Distinguished Colleagues Volume III" (Sheffield Labs LAB-1 SL5/SL6) was

PHOTO 7: Tubes and packaging.



scarily realistic and the TechTubes gave one of the finest reproductions of this LP that I have ever heard. Similarly, with the vocals of Thelma Houston's "I've Got the Music in Me" on Sheffield Labs LAB-2 SL7/SL8, the overall balance, presentation, and realism of the sound was superb. Again, the performances were thoroughly enjoyable, providing a musical and tuneful presentation that is very easy on the ears.

It was also interesting to note that these tubes definitely run cooler than their conventional counterparts. Even after constant running overnight, the case of my phono amplifier was barely warm and you could easily grasp a tube with your fingers to remove it.

Now, a word or two about microphony. Blackburn MicroTech Solutions puts a "Don't flick the tube" warning in their literature! This is all about flicking the tube with the fingers and the ringing that this will inevitably cause. They go on to say that this effect is perfectly normal and is a characteristic of the design.

I was using the tubes in a phono stage to amplify the very tiny signals from a moving coil cartridge, so any microphonic effects would be very apparent. It was noticeable that the tubes, when new, were somewhat microphonic. However, this effect diminished remarkably during the running in period, which was quite surprising. In fact, when fully run in, the tubes exhibited significantly less microphony than my existing audiophile tubes—most impressive!

Retailing for around £30 per tube or £40 for a matched tube (about \$48 and \$64, respectively), the TechTube E813CC is certainly a winner. In the near future, the company is also planning to introduce the E812CC and E811CC tubes as replacements for the 12AU7/ECC82 and 12AT7/ECC81, respectively. Further downstream, Blackburn MicroTech Solutions intends to produce replacements for the EL34, EL84, KT66, and KT88 power tubes, and investigate other tubes of various types to assess their suitability for the new technology.

For more information, see their website at www.techtubevalves.com/. *ax*



PHOTO 8: TechTubes website.



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The Essex Echo: Audio According to Hawksford, Pt. 1



Malcolm Hawksford had his first design, a console tape recorder with pre- and power amp and build-in loudspeakers, published in the prestigious *HiFi News* in June 1963 at age 15. It was an early milestone in a life and career marked by contributions to advance the state-of-the-art in audio. It led up to the Audio Engineering Society's Silver Medal Award "for major contributions to engineering research in the advancement of audio reproduction" in 2006, and a Doctor of Science degree for lifetime research achievements from Aston University (UK) in 2008. Jan Didden visited him at his lab at Essex University and spent many fascinating hours talking about audio technology.

Jan Didden (JD): Professor Hawksford, you published your tape recorder design at the age of 15 (Photo 1). When did you start to become interested in audio, and why?

Malcolm Hawksford (MH): Oh, I think it started when I was 8. You see, my father, and indeed his father before him back in the 1920s, were interested in music reproduction; my father had this 50s Grundig tape recorder, type TK820-3D I think it was. I was also interested in making puppet theaters, especially the lighting, and used to build control systems by winding variable resistors on wooden dowels, batten lighting, and spot lights using baking powder tins. It taught me a lot about lamps in series and parallel and the losses that could occur in connecting wires (smile). I became fascinated by those wires and lights and how they worked. So I got to play with tape recorders and circuits and also obtained

Professor Hawksford has published several articles for the audio press under the cover title "The Essex Echo." The Echo was a local Essex newspaper published between 1887 and 1918.

Another tape man who believes in compactness is Mr. Hawksford of Shrewsbury, who is already a regular reader at the age of 15. Surprisingly, the equipment illustrated incorporates all of the following: Brenell Mk V deck with three heads, Stern type C preamplifier with correction for all three speeds, RSC A11, 12 watt amplifier, TSL Mk II tuner, and a monitor speaker. The cabinet which is Mr. Hawksford's own work is finished in Raxine, and overall results are claimed to be "most satisfactory"

Now we move across the Channel to a German reader, Mr. W. Maass of Hamburg, who has just about as international a collection



The neat compact unit containing Brenell tape deck, Stern Type C, RSC Amplifier and TSL tuner. Designed and built by A. Hawksford of Shrewsbury. Aged 15.

PHOTO 1: The 1963 article in *HiFi News* featuring 15-year-old Malcolm's integrated tape machine.

electrical parts and old TVs from my father to experiment with, as at that time he worked for an electrical wholesaler.

I remember my first tape recorder didn't want to record: I had measured its recording head resistance with a multimeter, and, unknown to me, the head had become heavily magnetized. But I eventually figured it out, and, after degaussing the head (care of a Ferrograph de-gausser), it actually worked quite well. I was also very fortunate that my mum didn't really object when I turned

my bedroom into a lab and the carpet developed a silver sheen of solder!

JD: The 60s were a great time for a youngster interested in electronics anyway.

MH: Absolutely. You could get parts and kits for almost any purpose. After building a few kits, I quickly started to develop my own circuits and topologies. I still enjoy circuit design. The way it works for me is that I sort of juggle or model the circuit in my mind, actually visualizing the circuit and various voltages and currents, before I put it on paper.

I also was lucky that by the time I was ready to go to university in 1965, the University of Aston in Birmingham offered one of the first electrical engineering courses that focused on what was then called "light-current electronics," as opposed to power electronics. When I graduated at the end of my third year with a first-class honors B.Sc., it was suggested that I stay on for research. I then applied, and received, a BBC Research Scholarship and did my thesis on the "Application of Delta-Modulation to Colour Television Systems" (available on my website).

Using the (then new) emitter-coupled logic from Motorola, I was able to get up to 100MHz clocks, and that was in 1968 mind you. This logic family had to be interconnected using transmission-line techniques with proper termination to prevent reflections! This choice of subject proved rather fortuitous, as it gave me a strong grounding in delta-modulation and its close relation sigma-delta modulation, technology that was later to have a massive impact on audio systems in the 1980s and 90s.

JD: Do you see circuit design as an art?

MH: Yes, I think it is to some extent an art, or a bridge between science and art, in the sense that you develop a “pictorial” solution without knowing exactly how you got there. It sort of develops itself. I have been doing circuit design most of my life and it has become a “sixth sense”; I’m thinking in circuit blocks, sort of. In those early days you would try out different topologies, thinking it through, and trying to picture the currents and voltages in your mind while trying to get to the optimal solution.

JD: Is there a personal style in circuit design? Is there a “Hawksford” style in circuit design?

MH: To a certain extent I think there is. Designers usually solve a circuit problem slightly different from each other, perhaps based on how they learned to solve certain problems earlier and probably also depending on their personality. If you are a digital designer, you might choose to plug some design spec into a program that puts it in an FPGA for you. Likewise, as an IC designer you may have a library of standard cells or modules that you can use to lay out your chip.

In each of these cases the designer seems a step or two removed from the detailed design, making it more anonymous, unlike an analog discrete circuit designer. That said, I think that also sometimes circuits are designed differently for other reasons than you might think. I firmly believe that if you design an amplifier, and you take care of both the critical factors and secondary effects, such designs will tend to “sound” very similar. . . hopefully implying the performance is accurate.

Now, of course, the topology isn’t all

of it. People often become preoccupied with topology, but there are many more issues required to make a circuit into a great piece of equipment. There’s the power supply, the grounding layout, EMI issues, the quality of the components, the wire used—they all contribute to the final result. So, when you get the topology right—that is, get it to converge in terms of stability and linearity and such—then the secondary factors become important.

Let me give you an example. Most designers are aware that you must avoid sharing supply return paths between power and signal returns. The power return current could cause a “dirty” voltage across the return path that couples into the signal circuit. Even if you use a series supply regulator, you can still have this problem with a rock-stable and clean supply voltage, because the harm is done through the return current.

Now, if you use a shunt regulator, the “dirty” current can be localized and kept from signal returns, and that offers a major advantage. If it still isn’t enough, you can use what is called an “active

ground” or “dustbin” where the supply return current is not returned to the ground common at all but disappears into another, separate supply system (Fig. 1A-C).

JD: What is your view on the desirability and usefulness of blind testing to rate the performance of audio equipment?

MH: Well, I think you must use some kind of objective form of subjective testing method to isolate differences between components. Many people do not realize that they have a sort of internal perceptual model that determines how they perceive the auditory input. That internal perceptual model not only takes into account the sound feed from your ears into your brain, but also how you feel, your expectations, how bright is the environment, how relaxed you are, and many other factors. So if your internal perceptual model changes due to those other interference factors changing, your perception of sound can change.

I recall occasions where initially I perceived a certain difference between cables, and then I repeated it the next

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day and my perception was often quite different. I think this was due, at least in part, to my changed internal perceptual model. So some kind of objective test is required, but that said, I'm not a strong advocate of ABX-style double-blind testing (DBT).

The limitation in sensing a change in sound can put us in an unnatural situation, and I'm not sure we then function so reliably or sensitively. Possibly a better approach is a blind method that allows a relaxed and holistic type of lis-

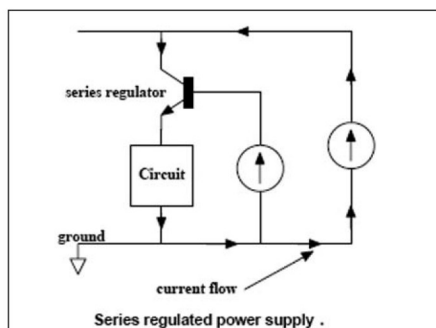


FIGURE 1A: Series regulator return current flow.

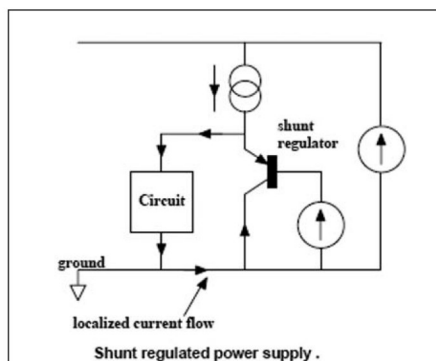


FIGURE 1B: Shunt regulator localizes return current.

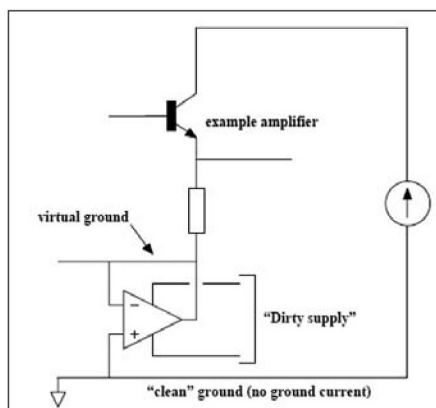


FIGURE 1C: "Active ground" dumps return currents in another supply.

tening session. Of course, with DBT it's easy to get a null result, so it may be a good method if that is your agenda.

My preference is to undertake listening tests in a completely darkened room. The fact that the equipment you listen to isn't hidden and could be identified with just a bit more light makes it much more natural and less stressful than being aware that the equipment is purposefully hidden from you. Also, being able to focus your senses purely on sound and not be distracted by uncorrelated visual input to the brain heightens your auditory perception. It is very easy to do and increases your sensitivity and acuity, especially in spatial terms.

In my experience it is not the same as closing your eyes. It seems that when you close your eyes when listening, you are sort of fooling yourself; it's artificial in a way and it still diverts some mental processing power away from your listening. You should try the dark room sometimes, although it's good to keep a small torch at your side!

JD: Another method correlating measurements with perception that gets some attention lately is trying to extract the difference between the "ideal" signal and the actual signal. In the past year I attended several AES presentations on systems to extract those differences and make them audible, such as the differences between unprocessed music and the MP3 version. Bill Waslo of Liberty Instruments, the makers of the Praxis measurement suite, even has a free version online (AudioDiffmaker).

MH: It is a very powerful technique which I explored formally in 2005¹, and we have employed the extraction of error signals over many years at Essex (see, for example, "Unification" articles on my website). The idea is that you have a system with both a target function (the design response) and the actual function with imperfections, so you can then represent the actual system in terms of the target function and an "error function." There's a lot to it, but as a simple example consider the frequency response of a high-quality CD player. You can assume that the target response here is a flat response to around 20kHz; if that is not the case, then, of course, you need to correct for the nonlinear target response

in the extraction of the error.

If, for example, there are response irregularities below, say, -40dB, it would give around 0.01dB frequency response ripple. You barely see that in the frequency response, as it's actually less than the graph line thickness! However, if you assume a flat target response, extract the error and then plot it on the same graph, which tells you much more.

Figure 2 shows the minute ripples in the response resulting from an imperfect DAC reconstitution filter. This tells you how far below the main signal you have some kind of "grunge" in the system, where ideally it should be below the noise floor. I like this type of presentation because it can inform you of the actual low-level error resulting from small system imperfections (both linear and nonlinear) that may cause audible degradation. This frequency response example is relatively simple, but in the paper¹ I give some examples of using MLS or even music signals to extract the low-level errors from ADCs and DACs.

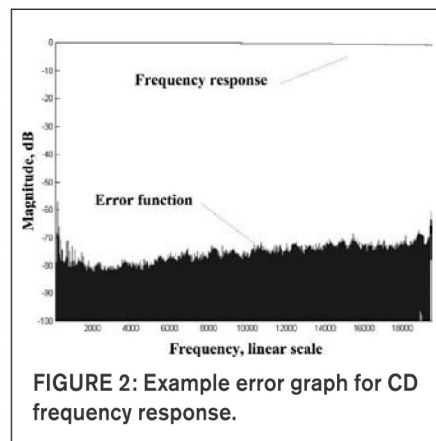


FIGURE 2: Example error graph for CD frequency response.

JD: If you can extract the error, can you then not compensate for it? Sort of "pre-distorting" the signal with the inverse of the error function? Possibly digitally?

MH: Well, compensating analog systems with numbers becomes complicated pretty fast. Most of these errors are dynamic or may arise from some interference of some kind, and although you can measure them accurately, you cannot predict them to any accuracy. The errors vary a lot with time and temperature and what have you.

It's been tried with loudspeakers, where you can develop a Volterra-based model to describe cone motion, for in-

stance, and use inverse processing to linearize it. But it is extremely difficult to keep the compensation model synchronized to the instantaneous cone position and movement. If you're just a little bit off, the results may be worse than without correction. It's much easier and cheaper to design a better driver!

There are some other techniques. There's a guy called David Bird, who used to work for the BBC and was using a current drive technique. One of my ex-Ph.D. research students, Paul Mills (who is now responsible for loudspeaker development at Tannoy), and I have also done some work on that subject. With current drive, the principal error is, in fact, the deviation of the $B\ell$ -product of the driver.

You can therefore measure the $B\ell$ deviation as a function of cone displacement, and if you then monitor the cone position, you can apply inverse $B\ell$ correction such that the force on the cone is proportional to the input current. We actually developed a transconductance power amplifier to current drive a loudspeaker, with several error correction techniques included in the design². We solved the low-frequency damping problem in two different ways. One was to use an equalizer; you measure the hi-Q resonance and then preprocess the signal to obtain the required linear response.

The other approach was to wind a thin wire secondary coil onto the voice-coil former of the drive unit, just voltage sensing, and to process that signal and feed it back into the transconductance amplifier. There was some unwanted transformer coupling from the main voice coil into the sensing coil which we had to compensate for with a filter. But since the main coil was current driven, it didn't matter if it heated up, and since there was no current flowing through the sensing coil, it also did not matter if it heated up. It worked very well; I remember that even using current drive with a tweeter also significantly lowered distortion.

JD: It wouldn't help with things like cone breakup.

MH: No, it wouldn't. And it adds an extra layer of complexity and things that can go wrong. It is also only suitable for

active loudspeaker systems.

JD: One issue that turns up in your work again and again has to do with jitter in some form.

MH: Well, yes, because it turned into an issue after we got the CD from Philips and Sony, and after the first euphoric reports, many people realized that what should have sounded perfect didn't. A major cause was jitter, which hadn't really been considered in those early years, probably because jitter is an "analog aspect" of a digital system. It can also manifest itself in different ways; it can disguise itself like noise (random and relatively benign) or as a periodic disturbance related to power supply ripple or clock signals, which is more objectionable, or it can be correlated with the audio signal, which also can sound quite bad.

So just saying "jitter" is not enough; its effect depends very much on how it manifests itself. In fact, I produced a paper at one time in which I designed a jitter simulator that allowed one to compute specific amounts and type of

jitter, noise or periodic or correlated to the signal, and add that to the clean signal so you could listen to its effect (see sidebar *Hawksford on the Sound of Jitter*).

You can debate its significance, but at least you can point to a measureable and audible defect, whereas a traditional jitter picture with sidebands and what have you doesn't give you a "feel" for what it sounds like.

I did a study with research student Chris Dunn³ (not the Chris Dunn who has published substantial work on jitter) which showed that the jitter introduced by the AES/EBU (or S/PDIF) interface protocol even depends on the bit pattern—in other words, on the music signal itself.

For instance, when you listen to the error signal of the phase-lock loop (PLL) on the digital receiver, you can actually hear the music signal that was transmitted through that digital link! It is distorted, of course, but this was clearly an example of music-correlated jitter.

Now there are known engineering solutions to eliminate that jitter later on,

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but it doesn't always happen in equipment, so there is the possibility when you transmit digital audio through a band limited link (and it is always band limited), you can get correlated jitter just from that process. We also showed that if you code the L and R signals separately, invert one of them, and then send both over that interface, almost all of that signal-related jitter would disappear. But it wasn't picked up on; such is the law of standards!

JD: How would you design the “ideal” DAC?

MH: The DAC chips themselves nowadays are very good indeed. Where you see the differentiation in quality is in stages like the I/V converter, a seemingly innocent subject. The sharp switching edges from the DAC output can only be perfectly reproduced with an I/V op amp that has infinite bandwidth and no limit on slew-rate. Any practical circuit will have nonlinearity and slew rate limits such that a transient input signal can slightly modulate the open-loop (OL) transfer of the op amp.

Modulating the OL transfer function means you modulate the circuit's closed-loop (CL) phase shift. What is interesting is that it looks remarkably similar to correlated jitter; they share

a family resemblance (**Fig. 3**). It also is similar to what people have been talking about as dynamic-phase modulation in amplifiers. Whenever an amplifier stage needs to respond very quickly, it tends to run closer to open loop and therefore is more susceptible to open loop non-linearity. So the I/V stage is clearly a critical stage, and although the underlying processes are different, the resulting signal defects may manifest themselves as correlated jitter, especially as the timing errors occur close to the sampling instants where signal rate-of-change is maximum.

Anyway, I really think we should not talk about phase modulation here, as that is more appropriate for sine wave signals and linear systems. We should talk about temporal modulation instead. There are many ways you can solve these issues once you understand them, possibly to design your I/V converter to be very wide band, or using a very linear open-loop circuit, or maybe some low-pass filtering between DAC and I/V stage. What you end up with⁴ is a discrete current-steering circuit that runs partially open loop and integrates the I/V conversion and low-pass filtering into one circuit, rather than bolting an I/V stage to a subsequent second-order filter as is normally done. Consequently,

you minimize the active circuitry involved (**Fig. 4**).

If you think about it, theoretically we are trying to make circuitry work flawlessly up to infinitely high frequency, which in principle cannot be reached. So at one time I thought maybe we need a totally different way to solve the problem of critical timing issues in DACs.

One possible solution I came up with was to modulate the reference voltage of an R-2R ladder DAC with a synchronized raised-cosine waveform. Rather than the DAC output staircase signal jumping “infinitely” fast to a new level at every clock pulse, it effectively made the new level the same as the previous and then ramped it up, so to speak, to the new level using a raised-cosine shape with the same period as the clock (**Fig. 5A, B**). Consequently, adjacent samples were linked by raised-cosine interpolation rather than a rectangular step function. This also helps a little with signal-recovery filtering. So, the rate-of-change of the currents coming from the DAC was dramatically reduced. I built a prototype to proof the principle and it dramatically reduced the timing and transient errors in the I/V stage.

In many ways I view I/V conversion after a DAC as the digital-system equivalent of a MC phono preamplifier.

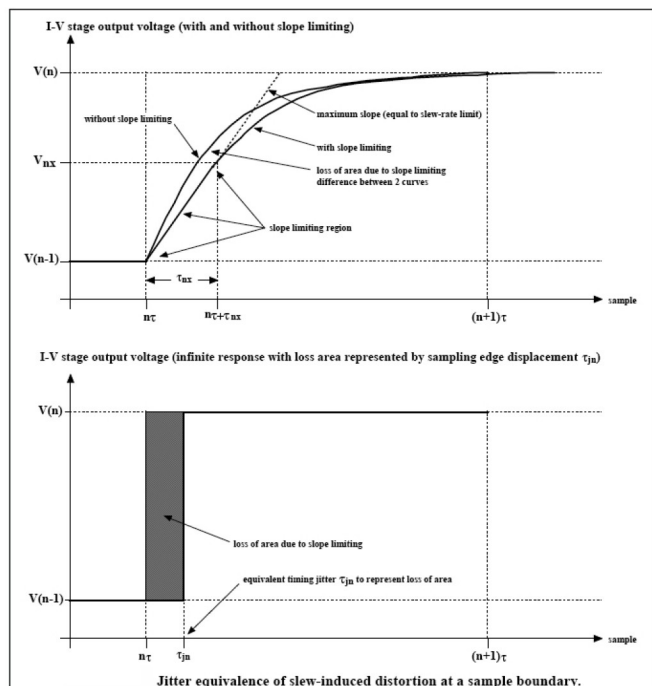


FIGURE 3: Slew-rate limiting in I/V converters has jitter equivalence.

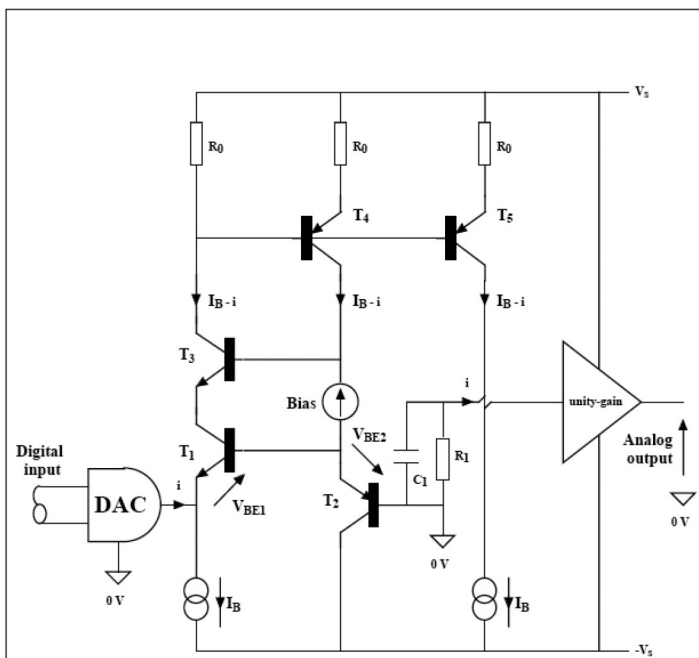


FIGURE 4: An open-loop I/V converter with integrated filter and input-stage error correction.

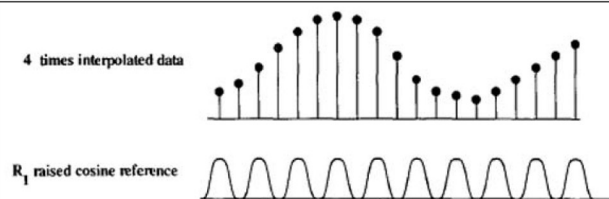


FIGURE 5A: Audio samples combined with raised-cosine DAC reference combine to...

composite signal
cosine interpolation

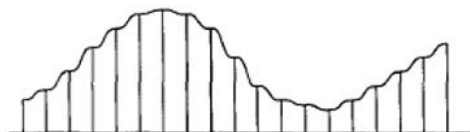


FIGURE 5B: ...dramatically reduce harmonics in analog output current reducing I/V slew rate requirements.

Although the application is totally different, I find that if you have learned to design a good MC preamp, that actually helps you to design a good I/V stage!

Another important issue is to locate the clocking source for the DAC very close to the DAC itself and slave everything, including the transport, to that clock. The clock should be free running, very pure and not controlled by a PLL; very often a PLL will only move the jitter to another frequency band and the frequency of oscillation is bound to wobble. There's nothing wrong with a free-running clock as long as you make sure that your data samples arrive on time, and you can do that with an appropriate buffer memory and data request protocol.

The CD player is a horrible RF environment, and you need to get the clock and DAC away from that source; just place a portable radio close to a CD player and do your own EMI testing! Even local supply bypassing of the DAC can couple noise into the supplies for the clock and increase jitter! So now you can list a few issues necessary to get it "right" in a digital playback system: the I/V; all the massive problems from EMI, supply, grounding, and so forth; and putting a clean clock right where you need it. I would speculate that if you gave a circuit topology to three different engineers to lay out a PCB and then build it, you would end up with three different results purely due to the differences in layout and component parts selection.

Now, how do you get a clean, stable DAC clock in your system? Suppose you have a transport and a DAC interconnected and you try to stabilize the DAC clock at the end of the digital interconnect; in principle you will succeed long-term, but in the short term that clock will wobble about and produce jitter. And even if you have your super

DAC with clean clock and PLL with low filter cutoff, you still are faced with an input signal that is not necessarily clean. It can induce ground-rail interference and your supply may become contaminated, so that incoming jitter may then bypass all your hard work and still end up affecting the output of the DAC. Memory buffers can, of course, help in the smoothing process, but beware of power supply and ground-rail noise.

JD: 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 inter-

face 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.

MH: Yes, network audio turns a lot of these issues upside down. It actually works the other way around. You put the DAC clock in charge, and it can be very clean and free running—no PLL—very low phase-noise. It is the way it's done in the Linn Klimax DS; the clock effectively "demands" audio samples from the network or NAS drive at its own pace to keep the buffer memory filled. I found the Klimax one of the cleanest and most articulate digital replay systems I've ever heard. For me, this is the way to go.

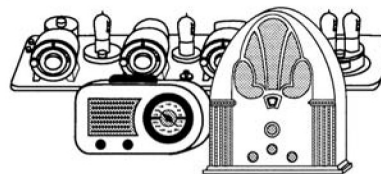
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lution 24/96 or 24/192 audio file, delivered through a top-notch network DAC, can sound absolutely stunning, and I have some wonderful Chesky recordings at 192/24. But even a 16/44.1 CD recording, when played through a network DAC implemented correctly, can also sound pretty spectacular. Maybe not quite as good as the hi-res stuff, but very, very good nevertheless, and you would be hard-pressed to hear the difference. Of course, the CD recording quality has to be first rate, but that's a very different story!

JD: There are several companies out there trying to make this happen. Mark Waldrep's iTrax.com allows you to download music on a pay-per-download basis, where the price depends on the quality. You pay perhaps \$2 for a 24/96 download, giving you actually the recording master, down to perhaps \$0.69 for the MP3 version of the same music.

MH: Yes that's an extremely good way to do it, and if you look at the Linn website you'll see that they offer similar services. Linn also gives you the option to buy their hi-res content pre-loaded onto a NAS drive, which for some people is more convenient than the hassle of downloading and setting up playlists on the PC. *Chesky Records* is also a very excellent source of music, and they actually have some 24/192 material. The B&W model that you subscribe to and obtain regular downloads is also interesting.

I think if you make the price right and especially if you provide high-quality recordings, people won't cheat, generally. And these specialty music providers also are extremely careful about the recording quality of the music they list, so that is one more uncertainty removed from buying a CD, where you may like the music but maybe the recording quality isn't so good. So to me it looks that networked audio delivery is slowly coming of age, yes.

JD: Can we spend a few words on loudspeakers and their part of the audio performance? In your keynote speech to the Japan AES regional conference in

2001⁵, you saw a great future to Distributed-Mode Loudspeakers (DML) to diminish the influence of room acoustics on music reproduction.

MH: Yes, indeed. You see, a DML has some great advantages. Rather than having a piston action like a traditional cone or panel speaker, a DML consists of a myriad of vibrating areas on a panel where in effect the impulse response of each of these small areas has low correlation with its neighbor. That is the significant thing which makes the polar response spatially diffuse. Now many people feel uneasy with that because it looks as though this will lead to a diffuse field, and it does!

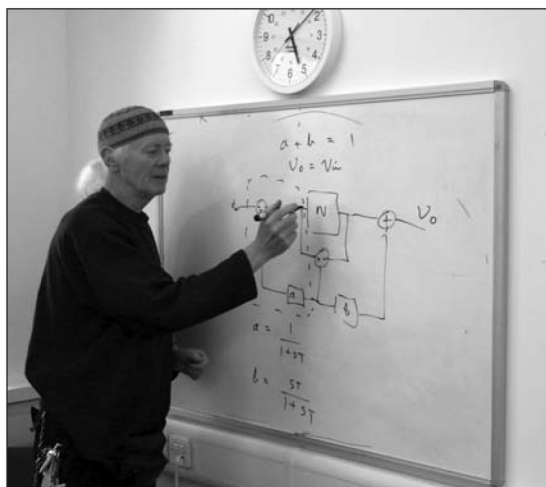


PHOTO 2: The Professor in his element: explaining feedback/feedforward concepts.

But, it does *not* lead to a significant breakdown of spatial sense or of instrument placing, because although the field becomes diffuse, the directivity characteristic does not. The major advantage is that the room acoustic reflections add with a significant degree of incoherence; they average out, so to speak, they are diffused. It helps to make an analogy between coherent light (from a laser) and incoherent light (from conventional lighting); in the latter the lack of interference results in much more even illumination without interference patterns.

To be honest, the sound stage itself does suffer a little bit, but the advantages can outweigh the disadvantages. You have no defined sweet spot, but you have no "bad" spot either when you move around the room. Furthermore, you could construct a DML as a flat panel, make it look like a painting, for

instance, which makes 5.1 or 7.1 surround so much friendlier in the living room! You could even make them an integral part of your flat-panel video screen or, in principle, weave them into the fabric of the room architecture.

But as far as I know only NXT has taken up the technology for use in specific circumstances, and successfully, I might add. Now, for regular stereo use, DMLs may not quite give you the sharp holographic image traditional loudspeakers can achieve, but in practice that will not often happen anyway. People seldom place their loudspeakers in the correct position in the room to realize the full potential for imaging.

Now that we are discussing the diffuse characteristics of DML loudspeakers, it reminds me of something similar I have done with crossover filters⁶, where the crossover transfer functions have a kind of random component added to them in the crossover region. The issue is: If you add the responses in a crossover on-axis, they add up and you should get a flat combined response. But if you add them off-axis, you normally would get a dip at the crossover frequency due to interference from non-coincident drivers. But with noise-like frequency responses, then for the off-axis sum, the interference is dispersed and the dip spreads out over

some frequency band around the crossover frequency and becomes less pronounced. You diffuse the problem, so to speak. It's similar to what DMLs do: I call them *stochastic crossovers* (Fig. 6A, B).

JD: You would favor active speaker systems?

MH: Yes. I believe that active loudspeakers have a number of advantages due to using separate amplifiers for each frequency range. Intermodulation, either directly or through the power supply, is much easier to avoid, as different amplifiers handle different regions of the audio band. Amplifier peak power requirements are also relaxed, in turn making it a bit easier to build high-quality amplifiers. And, assuming close proximity between amplifier and its associated driver, then those pesky loudspeaker cables are largely removed from

the equation (smiling).

In the 80s I consulted on what I believe was the first digitally corrected active speaker, developed by Canon. In fact, Canon funded a research project at Essex where we (that is, Richard Bews, now proprietor of LFD Audio, and me) produced a system that was ultimately demonstrated at their research facility in Tokyo. [JD: I have listened to that system in a large room in a General's castle in Belgium in 1984 or thereabouts; it left a vivid memory!]. There are two key aspects to digital loudspeaker processing. First, you can use it as a digital crossover filter, which will allow you to very easily correct any loudspeaker response errors as part of the crossover code. This is much simpler and less costly than using high-quality analog crossovers.

But once you have the capability, the urge is often to use it for room correction as well. I'm not a fan of that, simply because (ideal) room correction can typically be done for only one specific listener location, the ubiquitous "sweet spot." At any other location, the response, including the phase response, goes down the drain. The problem is very much wavelength dependent, so accurate correction tends to be limited to low frequency with less precise frequency shaping being applied at higher frequency. So, I'd use digital loudspeaker processing only for crossovers and loudspeaker correction. You should deal with room influences (other than low-frequency modal compensation) through other methods, where intelligent loudspeaker placement is one powerful way to improve your stereo reproduction.

Now, there's another aspect to digital loudspeaker equalization. Loudspeakers are a bit like musical instruments really; they have their own coloration and character where often you chose what appeals to you. Now if you equalize that loudspeaker, you may compromise the attribute that you liked, so, although being

more accurate you may have the impression something is missing or wrong. You should therefore consider digital loudspeaker correction an integral part of the design, just as is a passive crossover.

You shouldn't try to play with the correction or have switchable multiple corrections, just as you wouldn't want switchable multiple passive crossovers (apart from maybe some slight level correction in the low- or high-frequen-

cy range).

JD: That's the philosophy of AudioData in Germany. They sell one of these digital speaker/room correction systems. They will come to your home and set the system up to your liking with the corrections and all. They do not encourage you to play around with it. They put your particular correction files on the Internet, so when something goes wrong in your system you can download and re-install them. But in a practical sense it is a one-time thing—to your room, your speakers, and your taste, if you will.

MH: Yes, that's sensible. People should listen to the music, not to their loudspeakers or correction processing! There's one more thing I'd like to mention about placement. In the past, I have worked closely with Joachim Gerhard (founder of loudspeaker company Audio Physic in Germany), who came up with one of the best placement schemes I know. You need to avoid reflections coming from the same di-

rection as the direct sound, because this distorts your spatial perception (it messes with the head-related transfer functions we use in sound localization). Joachim drew an ellipse that just touched the inside of the room boundary. You then place the loudspeakers at the foci of this ellipse and place yourself at the middle of a long wall boundary (Fig. 7).

So, not only are the reflections now remote from the direct sound direction, they are also separated more in time, where both these effects have a major impact on localization and perception of the recording venue acoustics. In the context of a high quality two-channel audio system (using Audio Physic loudspeakers), it achieved one of the finest stereo soundstages I have ever heard. The sound seems to hang in there between the widely spaced loudspeakers; you can hear all detailed venue acoustics, very convincing, especially when

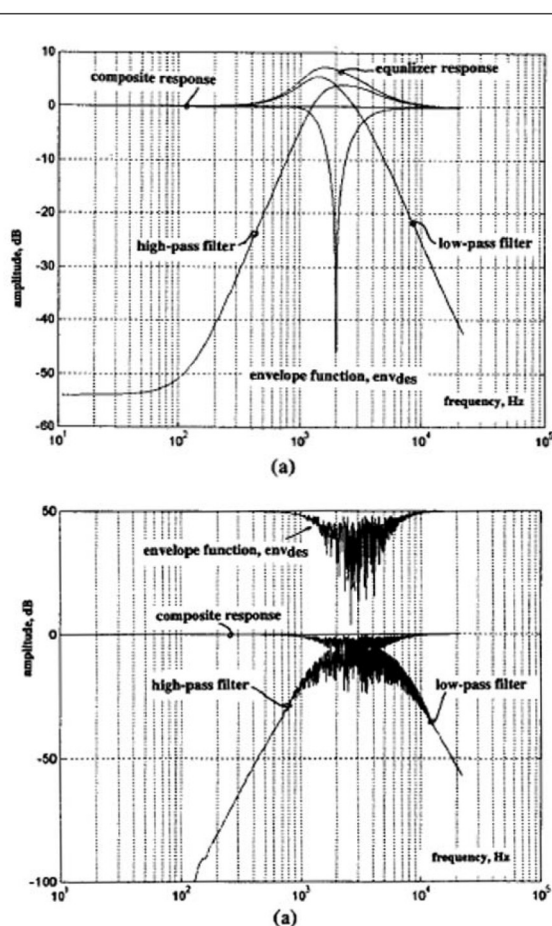


FIGURE 6A/B: Stochastic crossover has randomized filter characteristics that diffuses the off-axis crossover dip.

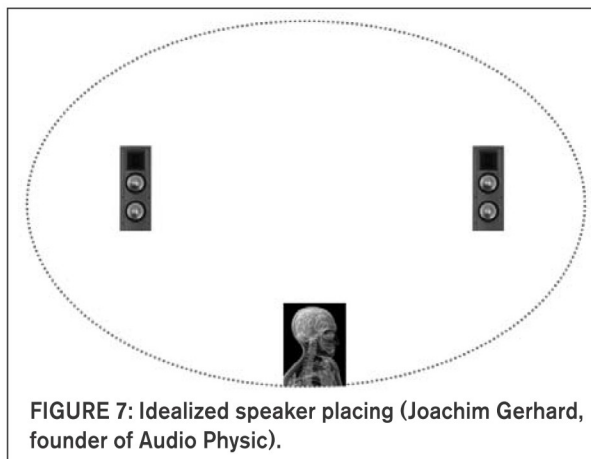


FIGURE 7: Idealized speaker placing (Joachim Gerhard, founder of Audio Physic).

the room is darkened! Also, having the loudspeakers widely spaced increases the difference signal between our ears, which helps to produce a more 3-D like image. Very interesting.

JD: Siegfried Linkwitz makes the point that you should place the speakers such that there is a minimum of 8ms temporal separation between direct and reflected sound, so that your brain can separate out the recording venue acoustics from the room acoustics.

MH: Yes, I very much agree with that. With most stereo placements, you add room reflections to the sound which “dilute” the spatial properties. So you may think that you have a larger image, but that is because it is blurred! The ellipse-based placement I just mentioned separates the direct and reflected sound both in direction and timing, and so helps your brain to keep the original spatial properties intact.

JD: And then there’s the issue of the speaker cables. I remember this paper you wrote in Marrakech I believe.

MH: I wrote a lot of papers in Mar-

rakech! I like to get away now and then to a quiet place, away from daily distractions. I would get up at 5:00 AM and then work for three or four hours. Those hours can be very productive, what you would call “quality time.”

But you probably refer to my article on cable effects and skin depth⁷. That one attracted a lot of criticism, and although there was a degree of speculation in it, I stand behind the major conclusions to this day. If I write something like that I always try to indicate what is fact, as we electronic engineers understand it, and what is more of a gut feeling.

In that article I addressed the topic of skin effect in the context of audio; however, it seems what I said was widely misunderstood and misquoted. Say you have a coaxial cable, consisting of lossless conductors (i.e., zero resistivity). All AC-current would then flow only on the two opposing inner surfaces as electromagnetic forces would push the charge carriers away from each other; the current would not penetrate the conductor and skin depth would tend to zero.

Here all the electromagnetic energy would flow only in the dielectric space

between the two conductors, propagating in an axial direction along the cable close to the speed of light with the conductors acting as guiding rails. Here the electric field is radial, while the magnetic field is circumferential with power flow in a direction mutually at right angles to these two fields that is along the cable axis. Now because all practical conductors are lossy, you inevitably get potential differences along each conductor, and this means that at the cable surface there must be a component of the electric field in an axial direction; however, the surface magnetic field is still circumferential.

When you consider these two fields, the direction which is mutually at right angles is now directed in a radial direction into the interior of each conductor. As a consequence, there is a propagating electromagnetic wave (loss field) within the conductor itself. Think of it as energy spilling out into the guiding rails which are now partially lossy and therefore must dissipate some energy.

When you solve Maxwell’s equation for propagation in a good conductor, you obtain a decaying wave because some energy is converted into heat. Also, the velocity is very slow and frequency dependent. It is this slowly propagating wave that determines the internal current distribution in the conductor and is the basis of skin depth; it also explains why skin depth increases with decreasing frequency. The “loss field” is at maximum at the surface and decays exponentially into the conductor.

So your current is no longer confined to the conductor surface but penetrates into the conductor; it depends on frequency and decays exponentially. Therefore, when you consider the series impedance of a cable, you find it is made of two principal parts. There is the inductive reactance due to the magnetic field within the dielectric between the conductors, and this, as you would expect, rises as 6dB/octave. However, the magnetic flux trapped inside the conductors has both a resistive and an inductive component. If the skin depth is such that the current has not fully penetrated all the way to the center of the conductor, then this component of impedance approximates to 3dB/octave.

What happens in practice depends

HAWKSFORD ON THE SOUND OF JITTER

There is a lot of talk about the effect of jitter on reproduced music. To help people to get a feel for it, I prepared some test files with well-defined amounts of jitter. Basically, what I did was to calculate the variation in digital sample values when a specific jitter signal would be present, and alter the samples accordingly. The tracks are on the *audioXpress* website and can be listened to or downloaded for your own use. Those of you adventurous enough to go through the details are referred to the reference below.

Track 0 is the original music, and the following tracks are the resulting amplitude-normalized “distortion” or error signals resulting from the types of jitter as listed:

Track 1: TPDF (triangular probability distribution function) noise-based jitter

Track 2: 2 equal-amplitude sinewaves (44100 - 50) Hz and (44100 + 50) Hz based jitter

Track 3: 3 sinewaves 50Hz, 100Hz, and 150Hz, amplitude ratio 1:0.5:0.25 based jitter

Track 4: sinewave 0.2Hz based jitter

Track 5: sinewave 10Hz based jitter

Track 6: 3 equal-amplitude sinewaves 1Hz, 50Hz, and 44100/4Hz based jitter

Track 7: All of the above 6 jitter sources combined.

NOTE: In a real-world situation these error signals require amplitude scaling to match the system jitter level; they have been normalized here to allow them to be auditioned.

Enjoy!

Reference: Jitter Simulation in high-resolution digital audio, Presented at the 121st AES Convention, October 5-8, 2006, San Francisco, Calif.

on the actual cable geometry and therefore which aspect of the impedance is dominant. I could go on, but I suggest you download "Unification" from my website for more information. So to conclude, at lower frequency the penetration is deeper while as frequency rises, the internal conductor impedance increases as the current becomes more confined to the surface layer, just as it would be if the conductor was lossless to begin with.

I also put some numbers to it and it turns out that when your conductor diameter is less than about 0.8mm, there are almost no skin effects even up to 20kHz.

Now, going back to loudspeaker cables, ideally you would want them to have just a very low value of resistance over the audio frequency band with no reactance. Due to the phenomena described above, that may not always be true, but there lies the art of loudspeaker cable design!

However, in understanding the problem with loudspeaker cables that can impact their perceived subjective per-

formance, there is another important factor. Even if cables are completely linear, they still feed loudspeaker systems that offer a nonlinear load due to drive unit impedances changing dynamically with cone displacement, suspension nonlinearity, and possibly saturation effects in crossover components. As a result, the current entering the loudspeaker is a nonlinear function of the applied voltage; this, in turn, means that any voltage drop across the (even perfectly linear) cable also has a nonlinear component which must be added to the loudspeaker input voltage. It is interesting to audition these error signals in real-world systems where distortion can be clearly audible. So in this sense cables do impact the final sound where this process is probably responsible for perceived differences in character or coloration.

aX

This interview with Professor Hawksford continues next month.

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Distortion Meter

Learn what this author discovered about audio connections while building his audio interconnect tester.

Brian, who hangs around my shop, tells the story of the first time he made a batch of guitar cables. Some of them sounded fine, while others did not. He had just discovered cold solder joints. After he re-soldered the bad cables, some of them still did not sound right. He finally replaced the plugs to get the high-end brilliance that his ears told him to expect. He has been curious since then about why bad soldering had appeared to have damaged a plug.

Soldering today is even more difficult with the effort underway to remove hazardous materials from the environment. Much solder used today is lead-free. I decided to test the different types of solder to see which one works and sounds best to me.

YELLOW SPEAKER

There seems to be controversy over what testing means in audio. Measuring a difference does not mean you can hear it. For example, if you paint your loudspeakers yellow, it would be a great surprise to hear a difference in a blind test. You could certainly measure and document the change in color. If you asked a panel of listeners to indicate their preferences, there would probably be a significant difference of opinion about the sound quality if they were looking at the loudspeakers.

I personally would ignore this data. Even with identical components, there are differences. The evidence that color changes sound quality would need to be almost overwhelming before it would even be worth considering.

On the other hand, if my ears tell

me there is a difference between two audio chains and I cannot measure it, that does not mean it is not there. The color change in the previous example probably is the wrong method to determine audio differences. I could measure frequency response, distortion, or other parameters to find the difference. Even if all the measurements were exactly the same, but my ears told me there was a difference, then it would be a question as to where my measurements failed or whether there was something else that should be measured.

TEST SETUP

Starting with a simple experiment seemed the way to proceed with investigating solder. I used a setup with two lead-free printed circuit boards and two matching paths, pads connected by 54 jumpers. The jumpers were made from the cutoff tinned copper leads of lead-free precision resistors. Low temperature coefficient resistors also have low coefficient leads!

I soldered one set of jumpers with Kester .031" 62% tin 36% lead and 2% silver rosin core, a second with Kester .031" 63% tin and 37% lead rosin core, the third with Radio Shack .032" 60% tin 40% lead rosin core, and the final path with Radio Shack .032" 96% tin 4% silver rosin core solder. All rolls were new and clean, the idea being to keep everything the same except for the solder used. I cleaned the variable temperature iron tip and re-tinned it with the solder under test before making the joints.

I did all the joints myself. For the lead-free solder, I turned up the heat a bit and got nice smooth melting and

good flow. I noticed the joints changed color and lost shine as they cooled. This happened very rapidly compared to the solders I am used to using. Also, sometimes the flux was still smoking after the joint had solidified.

The problem with solder joints is greater at low signal levels. A cold solder joint may require millivolts before reaching the break-over point and allowing current to flow. That is why some find tapping a bad microphone sometimes will turn it back on. The higher voltage will temporarily punch through the obstacle.

Using an Audio Precision System 2 to sweep each solder test path for harmonic distortion showed all paths were as good as the instrument. To make things more difficult, I lowered the levels by feeding the joints under test through a 100K metal film resistor terminated by a 49.9 Ω resistor in the preamp. I inserted a battery-powered LM4562-based two-stage preamplifier with a gain of about 200 in the return path.

I attached the preamplifier with BNC connectors, while connecting the circuit card under test with low-end gold-plated RCA connectors. A single input fed both circuits, although with different 100K resistors. Each output was connected separately. I ran multiple sweeps for harmonic distortion to be sure that random noise would have less of an effect.

RESULTS

The first sweeps showed a major difference. To be certain what was being measured was accurate, I swapped the test leads and jacks around. Turns out my first measurement was not what it seemed to be. The difference measured

was accurate, but it was the difference in the contact of the RCA jacks. One jack allowed the test lead to insert fully, but the other, being slightly out of round, did not allow the same contact.

The difference was almost immeasurable, after I forced in the plug. There seemed to be a slight advantage to one of the solders, but not enough to be definitive. The difference between a very tight plug and a tight plug was at least ten times more of a difference than the 110 solder joints! Basically, all I was seeing was noise levels.

SETUP

The next step was to make an even more sensitive measurement. After some experimenting with measuring very low level distortion, I was building a digital scale. This required me to use a Wheatstone bridge and instrumentation amplifier. Applying these designs and using the Audio Precision to measure the difference between two cables showed much better results. So I decided to build a stand-alone instrument for easy use and have an output for a spectrum analyzer to give more detailed results.

Based upon the previous experiments, the method would be to measure the intermodulation distortion of a low-level signal. If you have two sine waves, one of a high frequency and one a bit lower passing through the same connection, you get two sine waves out, if everything is perfect. At times one sine wave may be going positive when the other goes negative. Regularly these two different sine waves will collide and cancel each other out. But as long as the circuit path is perfect, you just get two sine waves out.

If there is a small dead zone around zero volts, then this cancellation will occur longer than it should. This will appear to produce a new sine wave at the difference in frequency between the two test sine waves. The same can happen when the system clips or even is nonlinear. It also will occur at the sum of the two frequencies. The bad news is the smaller the error, the lower the voltage of the new frequency. That is why the spectrum analyzer by itself did not show the difference.

It is generally accepted that inter-

modulation distortion adds a grungy, low-frequency, muddy sound to music and conceals a great deal of low-level detail. To test your connections, you need the two test signals to be low enough that they will be the typical audio signal level.

Brian's experience indicates that he heard the effect at guitar levels, which is about 100mV. Also, note that the output impedance of a guitar pickup is low rising with frequency and the input impedance of a guitar amplifier is high. Classic models of guitar amplifiers have two inputs, one a megohm and the other 100K.

A good choice seemed to be to use a source impedance of 100Ω. This is about the output impedance of much consumer gear. This type of output is usually loaded by a 10K or higher impedance. Here the aim is to produce at least 100K or a megohm of input impedance.

What frequencies to send posed an interesting question. They should be close enough to each other to produce a difference error in a region

where hearing is acute. The frequencies should also be far enough away from this distortion beat frequency to be easily separated.

I decided to excite my test connections with signals around 100kHz. These are high enough that it is possible to discriminate between the test signals and the distortion or noise that they produce. It is also at a low enough frequency that there should be little worry about RF effects. Although this frequency is outside the normal audio band, there is energy like this presented to much audio equipment, and a difficult test signal makes measurement easier.

Ordinarily I would just look at the difference frequency. My earlier measurements hinted that this by itself was not enough. The noise from the test path seemed to be increasing in the presence of signal.

The normal noise can be the always present thermal noise, or flicker noise, which is inversely proportional to frequency, or it could even be outside noise sources being radiated into the system. Lots of other possible noise sources

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exist, but these are the big three. A noise that only occurred or increased in the presence of signal would, of course, be a form of distortion. If this noise varied with different test paths, you could begin to believe it was the device under test that was producing it.

To generate a low noise, low distortion, stable in frequency test signal should not be too hard. My choice is a transversal filter oscillator. I can use a crystal-controlled clock oscillator, shift registers, and resistors to produce my test signals.

A catalog search showed I could get clocks that after processing would yield 96kHz and 93.75kHz test signals. At this resulting difference frequency of 2.25kHz, there are good choices for op amps that are low noise and low distortion to use for the detector filters.

The crystal-controlled clock signal tells my discrete time, delay line when to shift the data, which is just the clock signal divided by the number of shift register stages. Either it is high (5V) or low (0V). Because the input signal is a binary divided square wave, the level is very stable. I use a common inexpensive counter IC—type 74AC4020—to give me my data signal.

This square wave feeds the delay line, which is made up of 74HC164N serial-in parallel-out shift registers. CMOS process logic tends to include very nicely controlled output impedances, and use almost no power.

OSCILLATOR CIRCUIT

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

Selecting the right resistor values gives me a good approximation to a sine wave. Note that the voltage reference is the actual power supply voltage. Here I use a common 7805 type regulator. My experience tells me that although many manufacturers make them, they are not

all equal. I will not use the cheapest part here. By using a clean 12V source to feed the regulator, the regulation and noise seem to be adequate.

If you use all CMOS logic devices, much of the current draw is used by the regulator chip. Both my oscillators combined drew only 45mA.

The classic transversal filter works by multiplying each output by a carefully calculated constant to produce a filter. Because the resistor values are only within 1%, it seems reasonable to use around 100 steps for the filter. My good binary choice was 128, which was easiest to implement.

The results of all of these multiples are summed to form the correct value (voltage) by a single op amp. The result should be a sine wave made up of 128 segments.

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

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

Figure 1 is the actual oscillator circuit. You will need two of these.

COMPONENTS

I used a golden oldie for the op amp—an LM318, which has the slew rate to work for these levels and frequencies. One limit of this op amp is the power supply rejection ratio (PSRR), which is typically 80dB.

Each oscillator was fed through a low noise, low distortion 1K resistor to a

common 100 Ω mixing resistor giving a source impedance of 80 Ω . This is close to some guitar pickups and to the 75 Ω design impedance of an RCA connector. At this point you already have some distortion just from the very small bit of nonlinear or non-ohmic behavior that is in all resistors. This is related to the temperature coefficient of resistance but is also influenced by other factors. The parts list shows the exact critical parts used for my meter.

A source impedance of 80 Ω also has an internal thermal noise source equal to the square root of a constant times the absolute temperature times Boltzmann's constant times the resistance times the bandwidth. The calculation at room temperature is $\text{SQRT}(1.61 \text{ E-}20 \times R \times \text{BW})$. This is often expressed as $\text{nV}/\sqrt{\text{Hz}}$. For our source this would be 1.14nV for a 1Hz bandwidth.

I chose a low-noise op amp—the National Semiconductor LME49740, which typically has $2.7\text{nV}/\sqrt{\text{Hz}}$ of noise around 2250Hz. At first I tried a higher noise op amp, which just did not work.

The random noise in a circuit adds up according to the square root of the voltages present. You can build a circuit with more than one amplifier on the input signal. All of the outputs will be summed together. The signals add as they are all in phase, with the noise following the square root rule. As a result, there is an improvement in the signal-to-noise ratio equal to the square root of the number of amplifiers used. With four amplifiers the noise drops in half.

The feedback resistors also make noise. So keep these values as low as possible, but not so small the op amp cannot drive them.

Another area of concern is removing any RF interference. I did not do this at first, and even some passing cars would cause the meter to clip. I used small film capacitors here along with a small inductor. I just wound 12 turns of 26 gauge high-quality magnet wire around a 1/8" Allen wrench. This was about the same size as a resistor when finished. The difference this filter made was quite impressive.

The first four op amps of U1 and U2 are just the first stage of an instrumentation amplifier. I used surface-mount

This filter stage is followed by a simpler second bandpass filter consisting

FIGURE 1: Transversal oscillator.

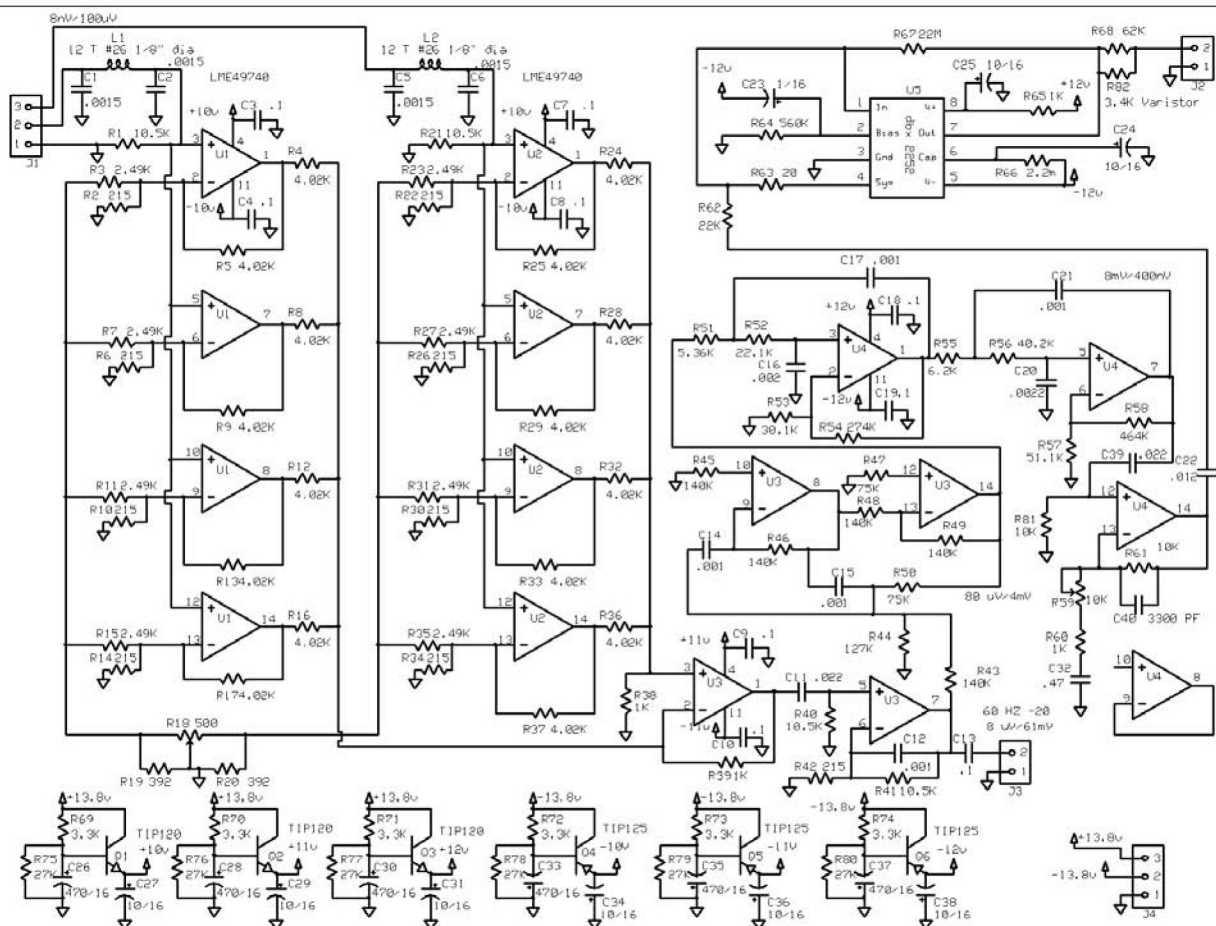


FIGURE 2: Analyzer section.

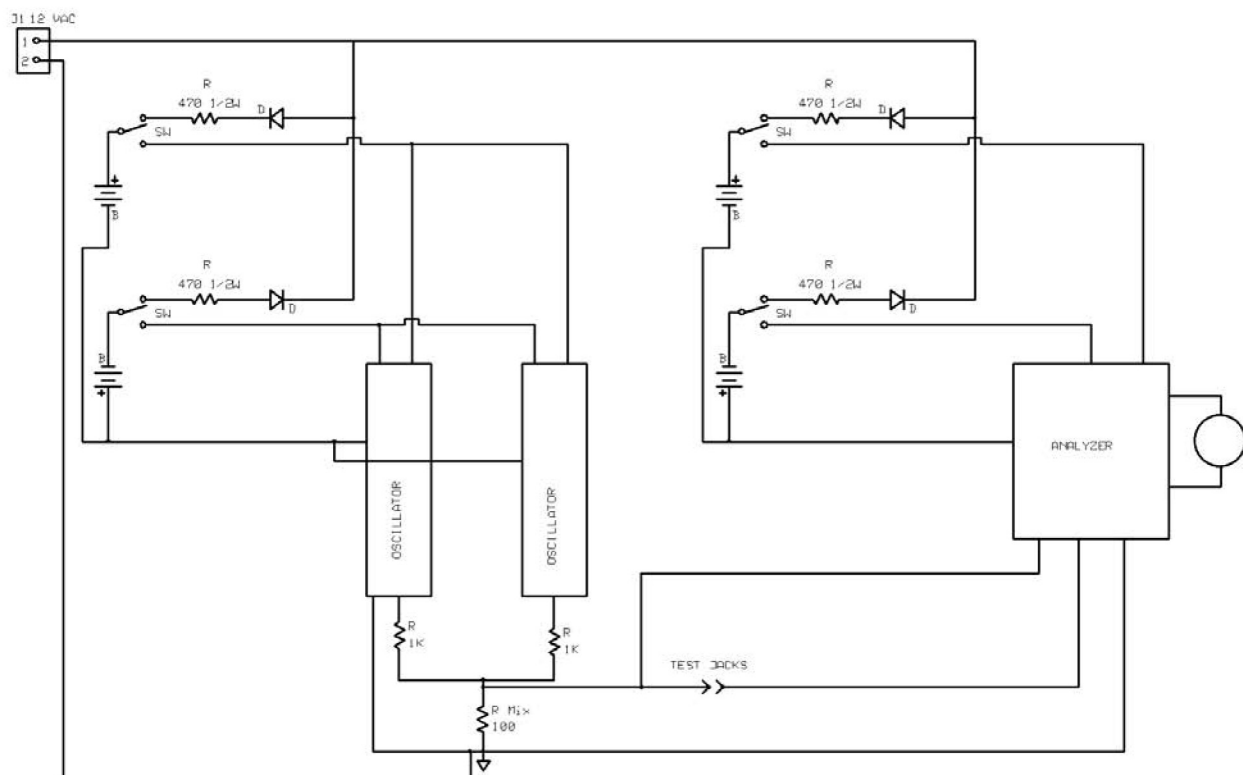


FIGURE 3: Circuit card interconnections.

of two more op amps in U4. This stage also provides about 100× gain. It is followed by a plain vanilla gain stage with just a bit of high frequency rolloff. This is where you can use any desired scaling. This is the last op amp used in U4 and provides voltage gain of 2 to 11×.

In the last stage a high-pass capacitor feeds a resistor used as a voltage to current converter. This feeds a THAT 2252 true RMS detector chip, which gives an output voltage of 6.2mV per dB. This can be positive if you are above zero reference level, or negative if you are below it. You can use R59 to adjust for an accurate zero reading. R68 and R82 convert the output voltage back into a current to feed the 50μA panel meter. R82 is a thermistor used to provide temperature compensation. The values used give a 25dB meter range.

I used an analog meter on the output. With the distortion component so close to the noise level, the meter swings a bit, and it is much easier to read an analog meter with noise than a digital one.

Because I went to a bit of trouble keeping noise out of the circuit design, it is vital to keep the power supply quiet. With the oscillator having a PSRR of only 80dB typical and the analyzer ICs only 60dB at test frequencies, this could be an issue. Batteries are among the quietest source. Lead acid is among the best of the batteries. Noise on the power supplies will show up in the output. I know! I used two completely isolated supplies. For the analyzer board capacitor multipliers, isolate the gain stages.

The only common connection is at the ground point of the mixing resistor. A trickle-charge circuit is active when you switch the meter off. I did not design this unit as a regularly used gizmo. A better charging system would be required if you plan to keep it plugged in.

A 16-gauge steel case housed my instrument. There are internal partitions shielding the oscillators from the detector amplifiers and the power supplies. Even with the shielding there is some leakage!

For the most sensitive uses you may wish to place the instrument and the test item inside a Faraday shield. For input

and output connectors I first used a 6-32 threaded brass rod in a plastic insulator. But later for ease of use I changed these to Neutrik RCA connectors.

To tune the meter you can use a short silver jumper and a spectrum analyzer. Insert the jumper and adjust R59 until the panel meter is at half scale. Then adjust R18 for a minimum reading.

Using the spectrum analyzer, see how many dB the spike at 2250Hz is above the background noise and adjust R59 until the meter reads twice that. (I did not replace the meter's scale, so 50μA equals 25dB of distortion above the noise floor.) If you do not have a spectrum analyzer, test all of your RCA cables and set the best one to read 10μA or 5dB—you will not be far off. **Figure 2** is the finished analyzer circuit.

LESSONS LEARNED

This instrument is difficult to build and make work reliably. I have provided parts lists. The ground wiring is most important to prevent contamination of the signals. My prototype showed me that some of the components I used were not good enough, so I needed to replace them. I used IC sockets on the first version. The second time I did not use them on the input ICs.

The ground wire that goes from the measurement card to the mixing resistor changed from 18 gauge wire to a piece of ½" wide copper foil. Finally, this became a piece of silver wire. The input leads are both silver wire the exact same length. One goes to the send jack, the other to the receive one.

Be sure to turn the meter on before you connect a new test cable. The turn-on spikes will change the path measurement if you do not do this! The op amps do heat a bit, so allow the meter five minutes of warmup to get more consistent readings.

As to the actual use of the instrument, I do not get a perfectly clean reading. The meter jumps a bit even when the connections are working well. Bad connections and cables give nice high readings. Computers, radio transmitters, passing cars, and just about any electrical noise source may cause the meter to jump. I use it on a wood table away from the AC outlets with

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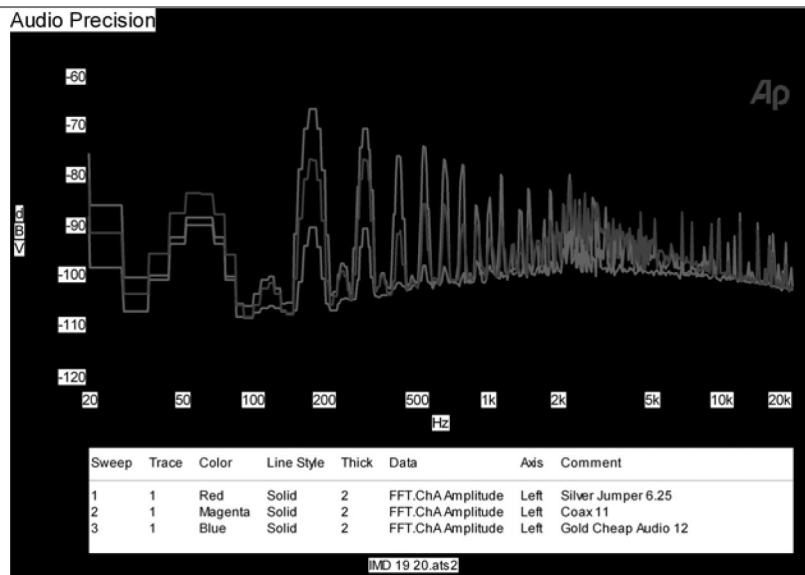


FIGURE 4: Comparing RCA cables.

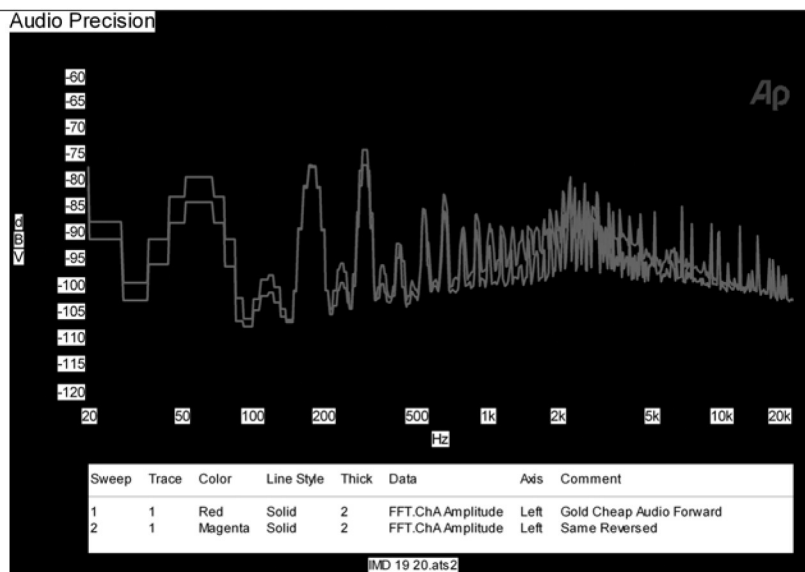


FIGURE 5: Same cable forward and reverse.

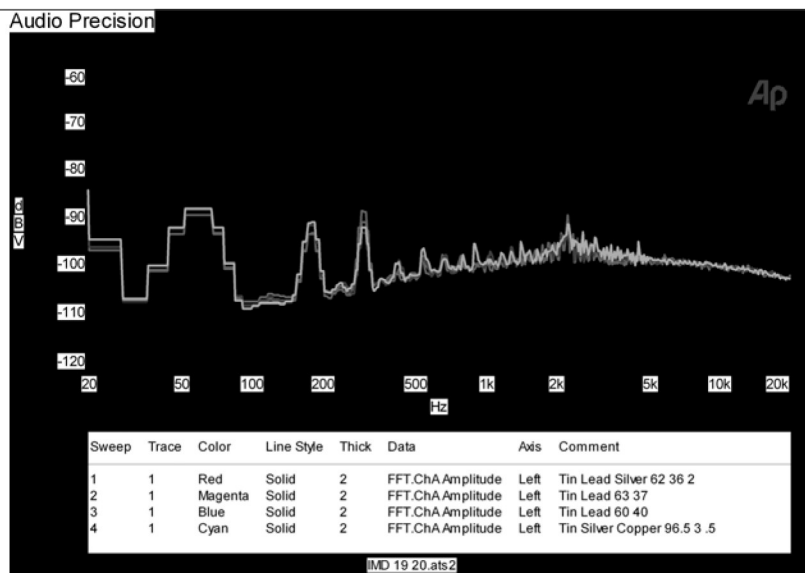


FIGURE 6: Comparing solders.

the lights out and wait for the meter to settle. With some patience you will find you can get repeatable and reliable readings. Looking at the results with a spectrum analyzer shows more insight into what is going on in the cables. Some have more noise, others more distortion.

If you build one of these testers, I hope you will find as I did that the readings do seem to correspond with what you hear. If you are curious as to some of what I learned, one of the first lessons was if you twist a brand new connector around after inserting it, the reading will go down.

That may explain why every time you connect a new piece of gear your system sounds better. A simple experiment you can try is to listen to your system, then unplug and re-plug everything five times. This should make your system sound better!

The second lesson concerns Caig Deoxit, which I have used to fix many noisy or even bad volume controls, touch switches, and even remote controls. I applied a bit to a noisy patch cable under test and nothing happened! I left to do a bit elsewhere and when I returned the noisy cable had become much better. So, Deoxit works, but it takes a bit of time to react!

I decided to test some inexpensive RCA-RCA cables I had on hand. **Figure 4** shows three different cables. One is a short silver jumper with decent gold RCA plugs. It reads 6.25dB above the noise floor on the meter and has the lowest noise and distortion.

The second is a video coax with cheap gold RCA connectors. It has much more noise and distortion and reads 11dB on the meter.

The last is a cheap audio cable with cosmetic gold-plated connectors. It reads 12dB on the meter. So there really is a correlation between how you build cables and the test results.

If you look closely at the test results in **Fig. 5**, you notice the effects of cable directionality. Sometimes, after taking a reading, I turned the cord around and read a different value on the meter. The spectrum analyzer shows why.

Surprisingly, audio cables really do work better one way! To see whether this difference really was there, I used

a two-cable switch on my test setup: a Sony CD player through a custom resistive volume control to my custom class A amplifiers into my corner horns. There was a noticeable difference in the apparent high end. Even my secretary noted one sounded louder than another! (Of course, I used the cable that tested for the most change.)

Here are some of the lessons learned:

- Solder connections were at least 50 times better than the best plain mechanical connectors.
- Gold connectors mating to gold connectors work best.
- Shiny (chrome, nickel?) connectors mating to gold are not quite as good—when clean and newly inserted—but become worse within hours.
- Shiny plugged into shiny is not as good as gold on gold.
- Some gold cords made for video and audio were horrible.
- Putting a surge through a cable lowers the distortion for a time, but it comes back. (Maybe the needle drop noise was good for hi-fi!)

As a test of these results, you may choose to replace internal connectors in your gear with straight solder joints, or at least put in a drop of Deoxit. IC sockets in your projects may not be a good idea; at the very least use the gold-plated ones. When doing listening tests, be sure you are not just listening to connector changes. I found I could hear differences of five units on this scale, which is less than the difference between clean and dirty connectors.

The distortion that is being measured is at low signal levels. Pro gear works at level of 0dBm, which is higher than consumer gear, meaning the percentage of distortion will be lower. The load impedance is also lower. You may prefer to try higher levels out with your gear and lower power amplifier gain.

The final results on solder are shown in **Fig. 6**. I just don't see any significant differences. 110 solder joints and it still is better than 6" of silver wire!

It may be too late for Brian, but I suspect that if he had cleaned his overheated connectors and removed the oxide that had built up on them, they may have sounded fine. *aX*

PARTS LIST

Part ID	Value	Part #	Supplier
C1, C2, C5, C6	.0015	P3915	Digi-Key
C3, C4, C7-C10, C13, C18, C19	.1	MKP1840410165	Mouser
C11, C39	.022	P3917	Digi-Key
C12, C14, C15, C17, C21	.001	FKP2-1000/100/2.5	Mouser
C16, C20	.0022	FKP2-2200/100/2.5	Mouser
C22, C32	.012	P3914	Digi-Key
C23	1/16	UVR1H010MDD1TA	Mouser
C24, C25, C27, C29, C31, C34, C36, C38	10/16	UVR1C100MDD1TA	Mouser
C26, C28, C30, C33, C35, C37	470/16	UVR1C471MPD	Mouser
C32	.22	P4667	Digi-Key
C40	3300pF	P3907	Digi-Key
J1	Solder direct		
J4	friction header	WM2701	Digi-Key
J2, J3	friction header	WM2700	Digi-Key
P4	terminal housing	WM2001	Digi-Key
P2, P3	terminal housing	WM2000	Digi-Key
P2 (2), P3 (2), P4 (3)	terminals	WM2312	Digi-Key
L1, L2	12T #26 1/8" diameter		
Q1, Q2, Q3	TIP120		Mouser
Q4, Q5, Q6	ZTIP125		Mouser
R1, R21	Open		
R2, 6, 10, 14, 22, 26, 30, 34, 42	215	PFC-W1206LF-03-2150-B	Mouser
R3, 7, 11, 15, 23, 27, 31, 35	2.49k	PFC-W1206LF-03-2491-B	Mouser
R4, 5, 8, 9, 12, 13, 16, 17	4.02k	PFC-W1206LF-03-4021-B	Mouser
R24, R25, R28, R29, R32, R33, R36, R37	4.02k		
R18	500	490-3020	Digi-Key
R19, R20	392	PFC-W1206LF-03-3920-B	Mouser
R38, R39	1k	PFC-W1206LF-03-1001-B	Mouser
R40, R41	10.5k	PFC-W1206LF-03-1052-B	Mouser
R43, R45, R46, R48, R49	140k	271-140K-RC	Mouser
R44	127k	271-127K-RC	Mouser
R47, R50	75k	271-75K-RC	Mouser
R51	5.36k	271-5.36K-RC	Mouser
R52	22.1k	271-22.1K-RC	Mouser
R53	30.1k	271-30.1K-RC	Mouser
R54	274k	271-274K-RC	Mouser
R55	6.2k	271-6.2K-RC	Mouser
R56	40.2k	271-40.2K-RC	Mouser
R57	51.1k	271-51.1K-RC	Mouser
R58	464k	271-464K-RC	Mouser
R59	10k pot	329*6W-1-103LF	Mouser
R60	1k	271-1K-RC	Mouser
R61, R81	10k	271-10K-RC	Mouser
R62	22k	271-22K-RC	Mouser
R63	20	CF1/4C200J	Mouser
R64	560k	CF1/4C564J	Mouser
R65	1k	CF1/4C102J	Mouser
R66	2.2m	CF1/4C225J	Mouser
R67	22M	CF1/4C226J	Mouser
R68	62k to 100k	271-62K-RC	Mouser
R69, R70, R71, R72, R73, R74	3.3k	CF1/4C332J	Mouser
R75, R76, R77, R78, R79, R80	27k	CF1/4C273J	Mouser
R82	3.3k thermistor	NC12MC0332KBA	Mouser
U1	LME49740		Digi-Key
U2	LME49740		Digi-Key
U3	LME49740		Digi-Key
U4	LME49740		Digi-Key
U5	THAT 2252L08-U		Mouser

TRANSVERSAL FILTER RESISTORS

Part #	Value	Part #	Value	Part #	Value	Part #	Value
R1	Open	R17	287k	R33	200k	R49	287k
R2	8.2M	R18	274k	R34	200k	R50	301k
R3	2.7M	R19	261k	R35	200k	R51	324k
R4	1.6M	R20	249k	R36	205k	R52	340k
R5	1.1M	R21	243k	R37	205k	R53	365k
R6	909k	R22	237k	R38	210k	R54	402k
R7	750k	R23	226k	R39	210k	R55	442k
R8	634k	R24	221k	R40	215k	R56	487k
R9	549k	R25	221k	R41	221k	R57	549k
R10	487k	R26	215k	R42	221k	R58	634k
R11	442k	R27	210k	R43	226k	R59	750k
R12	402k	R28	210k	R44	237k	R60	909k
R13	365k	R29	205k	R45	243k	R61	1.1M
R14	340k	R30	205k	R46	249k	R62	4.6M
R15	324k	R31	200k	R47	261k	R63	2.7M
R16	301k	R32	200k	R48	274k	R64	8.2M

Octode Amplifier

Can the pentode and triode live happily ever after in this new octo tube amp design?

Since the golden vacuum-valve age has come to an end, only triodes and pentodes are used for the construction of new valve amplifiers, all operating in accordance with the following modes: triode-ultra-linear-pentode. Along these lines, many engineers have designed and built the ultimate amplifier, only to discover a primarily new way of heating a room. Clearly, most new amps are just modern versions of those good old Telefunken 1960 amplifiers.

THREE MODES

At first all amplifiers were in *triode mode*. **Figure 1A** shows a PP (push-pull) triode stage. The balanced signal from the drivers modulates the triode, and the plate current variation powers the speaker. This system gives quite a good damping to the speakers, but the efficiency is low. A pair of EL34s, triode connected, will consume over 50W, but the output is only 16W, so the efficiency is about 30%.

The *pentode* is a current source with a high plate resistance. As you can see in **Fig. 1B**, here the screen grids are connected to the +HT supply. Without negative voltage feedback, this system gives no damping at all, but outputs much more power than the triode version. The same pair of EL34s in class AB delivers 35W, giving an efficiency of 70%.

Instead of connecting the screen grids to the plate or the +HT, some clever individual connected them to a tap on the primary winding of the OPT (output transformer), which resulted in the *UL mode* (**Fig. 1C**). As you can see on p. 4 of the 1957 GEC book *The Design of an Audio Frequency Amplifier*, sliding this tap from the center (0%) to the plate (100%) changes the pentode more and more into a triode. With this tap at 43%, the best results were obtained with respect to power output and distortion. Ever since,

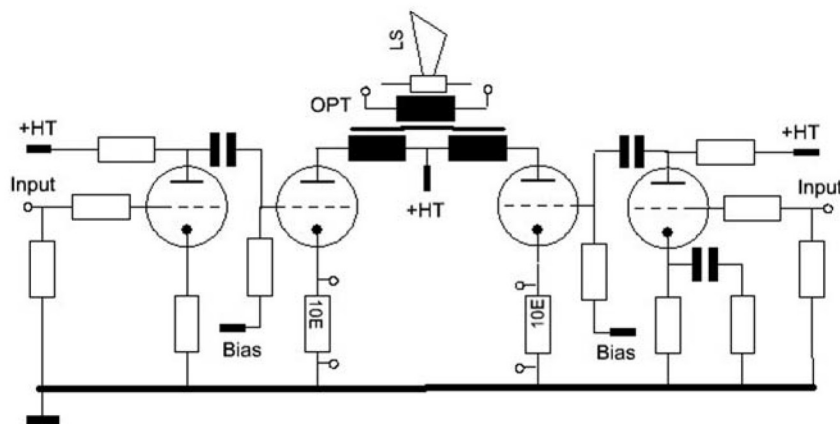


FIGURE 1A: Triode mode.

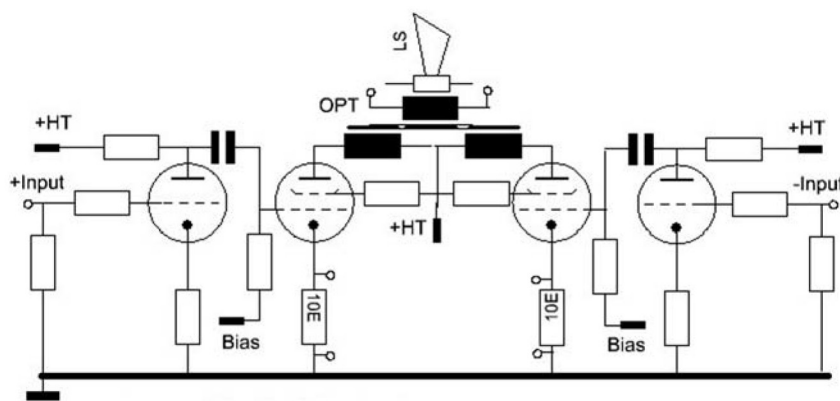


FIGURE 1B: Pentode mode.

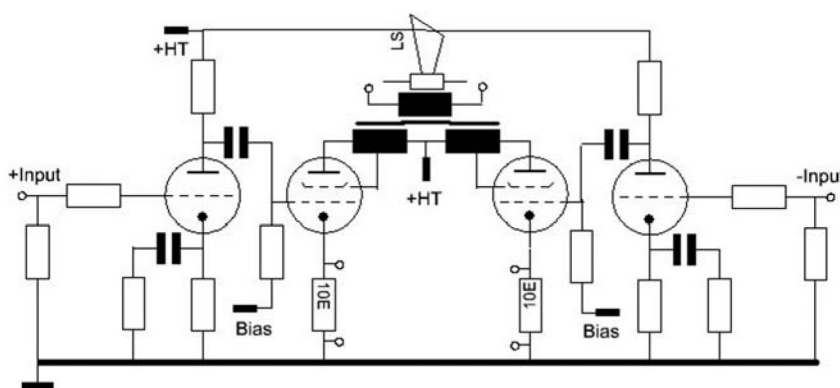


FIGURE 1C: UL mode.



PHOTO 1: Audio Aero SE monoblock. This big, heavy machine includes power supply (right) and the big triode 211 and tet-rode 813 (left).



PHOTO 2: Stereo SE Octode amplifier. In front you see the 6L6 assisting pentodes. With a switch you can engage either the 300Bs or the triode-connected KT88s. For this kind of amplifier a special OPT was made by Tribute Audio (pieter@tribue-audio.nl).

no significant progress has been made in the development of valve amplifiers. You can copy this book from Pete Millet's website. However, unbeknownst to most people, there is another mode.

EXTENDED CLASS A MODE

In this mode a pair of power triodes shares the speaker load with a pair of pentodes, each one in parallel with a triode but in class B mode, combining the well-known advantages of both triode and pentode. Some of these amps may have been made in earlier times, but—perhaps for commercial reasons—amps working in the UL mode dominated, but now the only company that still makes these kinds of amplifiers is Audio Aero in Toulouse, France (audioaero.com). They make a beautiful 40W SE amp with a 211 triode and a 813 tetrode, and a 60W P-P amp with a pair of EL34 triodes assisted by KT88 pentodes. Believe me, all these amps sound extremely well.

Many people may think that the characteristics of triode and pentode are mutually exclusive, but if you do it correctly the Ext-A mode works very well. The triode in class A addresses musicality and damping, while the pentode in class B delivers the extra power and increases the system speed.

Figure 1D shows the basic operation of the Ext "A" mode. Here the triode requires the full input signal, while the pentode gets only part of it, which should give the right amount of assistance to the class A working triode. Too much power from the pentode will over-modulate the plate voltage and consequently cut the top of the triode plate current, leading to a lot of distortion. Because both triode and pentode share a part of the plate load, each one will see a higher plate impedance than the nominal impedance of the OPT.

If the pentode gives twice the triode power and the OPT has an impedance of

3200Ω , then the triode sees a plate load of $3 \times 3200 = 9600\Omega$, and the pentode a load of $3/2 \times 3200 = 4800\Omega$. With this high plate load, the triode always works in class A, giving a low distortion, reasonable damping, and a moderate power output. When idling, there is a power consumption of $(2 \times 26) + (2 \times 10) = 72W$, and with an output of 50W it increases to about 110W. However, it would be nice if the pentodes also contributed to a low plate resistance of the system. In the Ext A mode, the triode and pentode may seem to be a happy couple, but they aren't really married. I solved this problem with the following system.

OCTODE MODE

Here the triode and pentode act like one big triode, but with eight electrodes, I call it octode mode. Figure 1E shows how it works. As usual, the input signal modulates the triode. The current of the triode is monitored by a small resistor in its cathode lead and amplified by a triode, which works as an I/V converter giving an input signal to the pentode that is a perfect copy of the triode current.

So the pentode delivers the same, but higher, plate current as the triode to the OPT. But beware of the phase; otherwise, a powerful oscillator will occur. By varying the signal to the pentode, you can control the amount of assistance the triode gets. Without a signal on the pentode, the triode will work in class AB, because the plate load of the OPT is relatively low.

Increasing the signal to the pentode increases the output as well. However, at

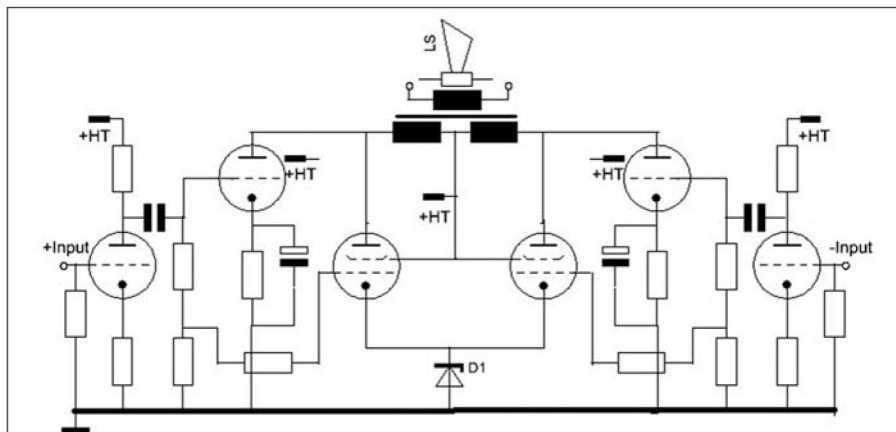
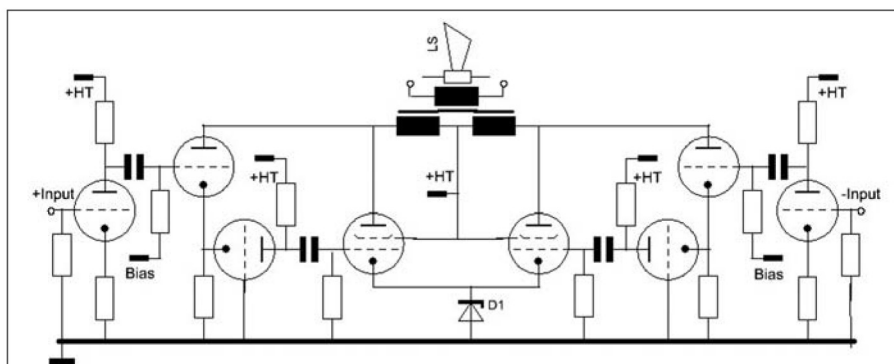


FIGURE 1D: Extended A mode.



a certain level the triode current variation will decrease, but this isn't a problem. The triode may deliver a little less power, but the system damping will be very good. In general the pentodes output two to three times the triode power. Less pentode assistance means less power but more of a triode sound. More assistance means more power and damping and a little more pentode sound.

The main advantage of this system is that without any negative feedback there isn't any crossover distortion but a very high damping of the speaker. Because any current variation of the triode is amplified by the pentode, the plate resistance is very low as well. Say the plate resistance of the triode is 900Ω and the total current variation is three times the current of the triode, then the plate resistance will only be a mere 300Ω .

DAMPING

So the combination of pentode slaved to the triode current leads to a large total current variation, consequently giving a rather low plate resistance and very good damping to the speakers. The couple works as a big single triode system but with far less power consumption than a multiple triode amplifier. This is quite like a happily married couple: the pentode is the spouse who is doing the hard work, while the other spouse (triode) keeps all well under control.

Regarding damping, the voice coil of the speaker has a rather high DC resistance, mostly about 50% of its nominal impedance. Even with an amplifier with zero output impedance, this resistor will always be in the speaker line.

I don't know why I so seldom heard a proper sound out of these famous high power, zero distortion, zero impedance amps. My son has Sonus Faber speakers

connected to all kinds of good amplifiers, from hybrid to MOSFET to tube. Now he has two Octode mode 60W monoblocks that I made for him, and he'll never give them back.

The guy that delivered a SACD to him could not believe his ears. After installing the Cambridge SACD player, he was very impressed and wondered how all this fantastic sound came from a valved amplifier.

The octode mode amplifier is nothing more than a class A triode amplifier assisted by a class B pentode amplifier sharing a common OPT (output transformer). A PP (push-pull) triode amplifier has a low efficiency and, even with the moderate power output, it still works in class AB. For example, two EL34s consume 54W, outputting 16W. The plate load is 5000 Ω , and the output impedance is 3500 Ω .

Consuming 80W, a pair of KT88s will output 30W with a plate load of 4k Ω and an output impedance of 2500 Ω . Without the use of feedback, none of these have good speaker-damping.

In the octode mode the two triodes are assisted by a pair of pentodes, not only to increase the power output, but also to lower the output impedance. Say the OPT has a primary impedance of 3200Ω

and the primary AC current is 100mA; then the primary AC voltage is 320V, giving a power output of 32W. With 10W output the triode stays in class A mode with a plate current of 32mA AC, so the pentodes must deliver $100 - 32 = 68\text{mA AC}$. This means that the plate load seen by the triodes is $10\text{k}\Omega$, and 4700Ω for the pentodes.

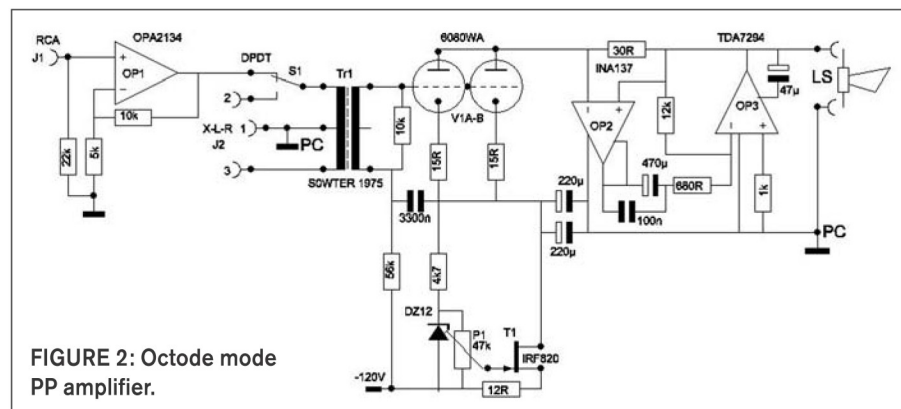
Being in class B, the peak plate current in the pentode is then below 200mA—no problem for a valve such as the 6L6. However, due to the assistance of the pentodes, the output impedance of the amplifier drops to $10/32 \times 3500 = 1100\Omega$. Measured on the 8Ω secondary, this means an inner resistance of $1100/3200 \times 8 = 2.7\Omega$. If the DC coil resistance of the loudspeaker is 5.3Ω , then the total DC resistance in the speaker line is 8Ω and the amp will act as a constant power source.

With a primary voltage of 320V AC, the plate voltage swings $\pm 230\text{V}$, and because the minimum plate voltage the EL34 needs is 150V, the high voltage under full load must be at least 380V DC. **Figure 2** shows the schematics of the amplifier, with its bias circuit plus a block diagram of the different stages.

STAGE 1

The first stage is a voltage amplifier with a 6922 double triode. Resistor R2 sets the input impedance to 47K, but you can change this value. The grid stopper R1 prevents oscillations and Rf signals from entering the amplifier.

Helped by the Miller-effect, the triode forms such an efficient high-pass filter that resistor R3 and capacitor C1 were needed to compensate for the loss in treble. The valve amplifies the input signal about 18× (25dB). The next valve



is just a zero gain phase splitter, coupled by the capacitors C2 and C3 to the next stage. . .

STAGE 2

Resistors R10 and R11 are grid 1 stoppers, and resistors R12 and R13 are the bleeders, connected to the negative bias circuit. The EL34s are triode connected, and resistors R15 and R16 connect the screen grids to the plates, which, in turn, are connected to the OPT primary. Resistors R8 and R9 connect the cathodes of the EL34s to ground, serving as plate current sensors and sending a current reference signal to the next stage. . .

STAGE 3

The third stage acts as an I/V converter. The AC signals from R8 and R9, being an exact copy of the plate current modulation of the EL34s, are amplified 25× (28dB) by the 6922 triodes and, via capacitors C4 and C6, sent to grid 1 of the pentodes in the next stage. . .

STAGE 4

Here R20 and R23 are the grid stoppers for the 6L6s, which are biased by the bleeders R19 and R22. The cathodes of the 6L6s are grounded by resistors R21 and R24 (12Ω), so the DC current of the pentodes can be set by measuring the voltage drop; 240mV is a plate current of 20mA in each pentode. The screen grids of the 6L6 pentodes are connected to the +HT by resistors R25 and 26.

The plates of the 6L6s are also connected to the primary of the OPT. Note that because V5-6 invert the phase of the signals coming from the EL34s, the plate connections of the 6L6 pentodes must be reversed as well. If you do it incorrectly, a powerful AC generator and maybe a damaged tweeter will be the outcome. I had quite good OPTs on hand but for this amplifier I ordered two OPTs from Peter of Tribute-Audio, surely the best you can get.

BIAS CIRCUIT

The -44V bias voltage comes from a single rectified 30V AC, and the 12V zener D3 drops it to -32V, which is about the maximum you need to set the 6L6s in class B. The 12V zener D2 again lowers the bias to -20V to set the plate current of the EL34 triodes. In general, when

the mains voltage varies, the plate current of V3+4 and V7+8 will vary accordingly, but this current variation is partly compensated by the change in the bias voltage. But there is still quite a high change in plate current left. In this bias system, a change in the mains voltage (hence the plate voltage) of 10% will vary the bias voltage 20%, leading to an almost stable plate current.

WIRING

Figure 3 shows the wiring of one channel of the octode amplifier. From left to right, you see stages 1, 2, 3, and 4. Using a pair of three-terminal boards will allow you to connect all parts to these TBs and the valve sockets. Use twisted wire pairs for the filaments (F).

As mentioned above, double-check the wires going to the OPT. This amplifier may look a bit complicated, but is quite easy to make if you view it as two different PP amplifiers on a single chassis. I suppose that for people making such an amplifier, there's no need to describe for maybe the thousandth time how to make a proper power supply. Just be sure that it delivers a HT (+Hv or plate voltage) of about 380V DC with a no-load current of 200mA and a full-load current of 350mA per channel.

The high voltage power supply I use myself is a bit smaller, but this only drops the maximum power out, not the performance. As you know, the filaments will at 6.3V AC consume about $2 \times (1.6 + 0.9 + 0.6) = 6.2A$. If hum is a problem, make a center tap on the 6.3V with two

220Ω resistors and connect this center to a small positive voltage.

STARTING UP

Putting this amplifier into operation is easy if you start up without the power valves in place and use a Variac to gently increase the incoming mains voltage. Be sure that both filament and bias voltages are OK. To limit the no-load plate voltage, don't turn the Variac all the way up. Check the input stage with the phase splitter for proper operation. Set all the bias voltages to their maximum negative value. Now insert the two EL34s and watch the plate currents.

After warming up the amp a bit, adjust the mains voltage and then the negative bias voltage for 2000mV DC on the cathodes, giving a plate current of 75mA. Connect an 8Ω load to the 8Ω tap of the OPT. Hook up your generator and adjust the output to 4V AC (2W).

Now the I/V converter should get a signal of about 400mV from the cathodes of the EL34s and amplify it to a signal of about 10V AC on the grids of the 6L6s (not in place). Remove the 8Ω load and note the increase of the output voltage. I measured 7.3V AC so the triode-only output impedance is $(7.3 - 4) \times 2 = 6.6\Omega$. Disconnect the mains, switch off the generator, and insert the 6L6 pentodes.

Power up and adjust the pentode plate currents to 20mA that is 240mV DC at the resistors R21 + R24. Reconnect the 8Ω ballast and adjust the input output for an output of 4V AC; remove the 8Ω ballast to check the no-load output volt-

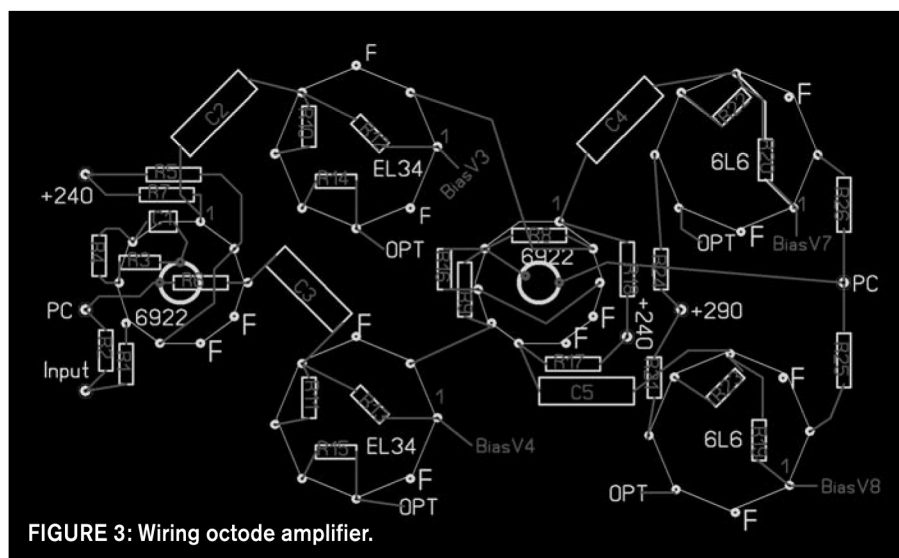


FIGURE 3: Wiring octode amplifier.

CONTRIBUTORS

Neville Roberts ("New Approach to Tube Design," p. 6), a graduate in Physics from the University of Bath, is a Chartered Scientist, Chartered Engineer, Chartered Physicist, and a Fellow of the IEE (now the Institution of Engineering and Technology). He joined the UK National Health Service (NHS) in 1992 having held various positions in the fields of science, electronic engineering, and information technology both in the Ministry of Defence and in the private sector. He is currently the Director of Information Technology for NHS Somerset. Neville has also been involved in the design of high fidelity audio equipment since 1975 and has been a freelance journalist since 2001 writing technical articles and equipment reviews, mainly for a UK Hi-Fi magazine.

Jan Didden ("The Essex Echo: Audio According to Hawksford, Pt. 1," p. 12) built his first OTL amp with 807 tubes 35 years ago. He has built speakers, preamps, and tape recorders, but is most interested in power amps, especially using error correction as discussed by Hawksford. Many of his projects have been published by *audioXpress*. Now retired from a career with the Netherlands Airforce and NATO, he tries to complete all those half-finished projects accumulated for lack of time. He now also has the time to travel to interesting audio events and interview audio luminaries. His projects are documented on his Linear Audio website.

Ed Simon ("Distortion Meter," p. 22) received his B.S.E.E. at Carnegie-Mellon University. He has installed over 500 sound systems at venues including Jacob's Field, Cleveland, Ohio; MCI Center, Washington D.C.; Museum of Modern Art Restaurants, New York; The Coliseum, Nashville, Tenn.; The Forum, Los Angeles; Fisher Cats Stadium, Manchester, N.H.

AJ van Doorn ("Octode Amplifier," p. 30) resides in The Netherlands.

Greg Koster (Review of Guru QM10 Speakers, p. 36) has written reviews for *The Sensible Sound*.

James Frane (Review: Lipinski L-707 Loudspeaker, p. 39) has written for *The Audiophile Voice* from at least 2002 to present, as well as *The Sensible Sound*.

PHOTO 3: PP octode amp under construction. In the center are the triode-connected EL34s and the 6L6 pentodes to the right and left.

age again. I measured 5.4V AC, which means that the output impedance has dropped to $(5.4 - 4) \times 2 = 2.8\Omega$. This is close to the estimated values, but can vary due to losses in the OPT.

MEASUREMENTS

The plate dissipation of the EL34s is well below their maximum of 30W, and for the 6L6s it's under 10W, so a long tube life can be expected—this is especially important with the new valves coming from the Eastern countries. I measured the distortion with a 4V AC output and found 0.08% 2nd and 0.01% 3rd harmonics for this octode mode amplifier.

Concerning the bandwidth I measured the following with respect to a 1kHz signal:

20Hz	-0.9dB	50Hz	-0.3dB
2kHz	0.0dB	20kHz	-0.1dB
40kHz	-0.3dB	60kHz	-1.0dB

aX

COMPONENT LIST

C1	= 15n
C2-5	= 1μF-Kimber 400V DC
C6, C7	= 220μ-elco 63V DC
D1, D2	= 12V zener
D3	= 1N4002
P1-4	= 10k
R1	= 1k
R3	= 1k2
R4	= 4k7
R5	= 82k 1W
R6, R7	= 27k 1W
R8, R9	= 27E
R10, R11	= 1k
R12, R13	= 270k
R14, R15, R25, R26	= 220E
R16	= 1k8
R17, R18	= 68k 1W
R19, R22	= 330k
R20, R23	= 1k
R21, R24	= 12E
R27	= 3k3
R29	= 47k
V1, V2, V5, V6	= 6922
V3, V4	= EL34
V7, V8	= 6V6
OPT Tribute Audio	40W
Prim:	3200Ω sec :8Ω

PHOTO 4: Left channel of PP octode amp, with OPT by Tribute Audio.

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Guru QM10 Speakers

by Gregory Koster



PHOTO 1:
Guru QM10.

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Price: (\$2395 for black and piano lacquer and info and matte black

Before General Motors bought Saab and denatured it, Saab automobiles had a devoted following based on extreme engineering aimed at producing the absolute best car for its intended purpose. They weren't inexpensive, and they couldn't do everything that a Jaguar or a Chevy Suburban could do—but on the rally circuit in the 60s and 70s they were the top guns and everybody else ate their dust.

Guru speakers are the 21st century equivalent of those early rally Saabs. Designer Ingvar Öhman has invoked the Norse goddess Sjöfn to marry physics and art in order to produce "the world's only known full-range mini-monitor." The Guru QM10 is a two-way, vented-port design that claims flat bass down to 30Hz from a bass driver with a 4.25" active membrane diameter in a ten liter enclosure

(12" W × 9" H × 10" D).

They also claim to avoid the narrow "sweet spot" limitations of many high-end speakers, aiming to produce a "holographic" image over a wide area of the room, and to do so in a typical listening environment (not the unnatural anechoic chamber of a certain other northern country's test lab). And to top it off, these qualities are said to result from speaker voicing designed for placement right up against the wall, which is how all mini-monitors should be designed. After all, if you have enough space to pull the speakers out into the room, you could fit a floor-standing cabinet and help physics give you full bass extension. [Note from Ingvar Öhman, R&D Guru Pro Audio: I do *not* recommend to pull the speakers out into the room, just because there is space enough to make it possible. But yes, the bass modules + QM10 can be used at a distance from the speaker wall, if there is something hindering putting the speakers close to a wall—or they can be close to the wall (still with the QM10s on top) giving an ever greater potential. So, having the speakers out in the room is an option (thanks to the controls on the bass module) but we still recom-

ment that it must be made on a computer-aided milling machine.

The upshot is a claim that the Guru QM10s are small, exquisite (in the optional piano lacquer) speakers that hug the wall and practically disappear on their diminutive 24" stands to achieve a high degree of spousal acceptance while still producing audiophile sound—"the Holy Grail of audio."

So, is this Granada I see or only Asbury Park? Let's crank them up and see.

SETUP

To be fair to the speakers under test, I always try to follow the manufacturer's setup recommendations as closely as possible. For a valid comparison I also position the comparison speakers in the same location, even though this may not be the position they were designed for. Thus any negative comments about the comparison speakers shouldn't be held against them—they may sound much better if used as their designers intended.

Lars Erickson of Sjöfn Hifi [no longer representing Guru audio] suggested putting the Gurus as close to the wall as possible (less than 6"), toed in so that you just see the outside edge of the cabinets,

"Guru speakers are the 21st century equivalent of early rally Saabs."

mend having them (the whole package) close to wall, to maximize dynamic headroom and minimize distortion (i.e., being as kind and respectful as possible to the music being reproduced).]

The midrange/woofer driver is Ingvar's own design, custom-made by Scan-Speak of Denmark. The Mylar tweeter is modified by Ingvar. And the cabinet is designed to such a tight specification

with any kind of foam on the wall toward the center, and with the tweeter just below ear level. [Note from Manufacturer: The figure 6" does not come from Guru. In fact the user's manual states *"very close to a damped wall."* By that we mean as close as possible, possibly only giving room for a 2", maximum 3" thick absorber behind the speakers' inner back corner.—Ingvar Öhman].

He also noted that designer Ingvar Öhman doesn't want subwoofer reinforcement. [Well, I haven't really said that. Subwoofer reinforcement *can* be a good way of improving the performance of a sound system, as long as it utilizes dual subwoofers that can work stereo-phonically, and is of a quality equal to the QM10's own in the bass range. I have, however, designed a bass solution for the QM10 that takes into account the small size of the QM10 and the possibility to have specialized bass modules (we prefer to call them that, not subwoofers) in close proximity to the QM10s. This also makes it possible to choose a different crossover frequency than the usual 80Hz. My experience is, however, that a subwoofer is not needed with the QM10 when played at levels most people would use in their homes. So, I do believe (i.e., my experience is) that most subwoofer systems do not improve the bass of the QM10, but there are exceptions. This is not a black/white question, but one with a grey answer. Very light grey yes, but still grey.—Ingvar Öhman].

Following this advice, I positioned the Gurus (and the comparison speakers) 61" apart center-to-center, toed in as directed, with the center of the back 3.25" from the wall (the distance ranged from ¾" to 6" at the edges of the cabinets). I placed couch cushions on the back wall from the inside edges of the speakers toward the center. I sat 66" away diagonally, so that I was well away from the back wall.

The only part of the instructions that I couldn't follow was the ear level. [That is (as said) not an instruction coming from me or from anyone else at Guru, so—no problem! Listening from any height is fine. At least ranging from 3" to 6"! The "magic" will be present at any height there within. (Magic meaning opening up a well working holographic window to the sound stage.) Actually, that is what holophony is all about—enabling people to be viewing the art, visual or sound, from a wide choice of positions in the room, horizontally, depth vice and vertically—just as in front of a real life music group!—Ingvar Öhman].

Lars had supplied 24" stands with the Gurus, but because the speakers are only 9" tall, it was impossible to sit low enough to position the tweeters "just below ear level." Given all the claims about designing these speakers for typical listening environments, I thought that I should sit in a typical chair even though that put my ears well above the tweeter levels. [Our recommendations (user's manual p. 6) are that the speaker should be placed at a height of 2' (60cm). The consequential height of the tweeter compared to the ears is not important and should not be used as a measure. Anything goes. The output of the speakers is timbrally modulated as a function of the listening height, to always produce a stable sound stage at a constant height above the floor, behind the loudspeakers, with all its intrinsic height preserved. This is one of the more important features of the speaker and a consequence of my ambition to make it create a holographic (holophonic) window into the world of music (or other sounds).—Ingvar Öhman].

My comparison speakers were Aperion Intimus 5B bookshelf speakers, and a ten-year-old pair of NHT Super One satellites paired with the NHT SW2Pi subwoofer, which has been the reference set in this system.

CRITICAL LISTENING

Dvorak—Serenade(Delos3011)—The Gurus were lovely on this recording, played at a chamber music level of 76dB (pink noise). The bass was full and balanced, although the deepest pizzicato notes were a little fainter than the rest; highs were not rolled off but also not screechy; the middle was balanced both tonally and on volume.

The Aperions were slightly warmer and had less extension on the highs.

The NHTs (sats and sub) were similar to the Aperions, except that the bass extended further down. Even without the highest extension they felt very balanced.

The Gurus with the NHT subwoofer sounded the same as the Gurus alone. The bass may have been slightly fuller or deeper, or I might have been fooling

myself (it was that close). The subwoofer didn't pay its way here.

On this well-recorded chamber music selection, the Gurus (sans sub) were the winner.

Rosemary Clooney—My Buddy (Concord 4226)—The Gurus were tonally cool but not forward. Rosie's voice cut through the orchestra but sounded normal (some speakers that place her in front of the orchestra sound as though she's singing into a megaphone). The snare and cymbals were prominent over the bass drum and the rest of the mix. Great definition—you could hear every instrument clearly. No hint of added warmth, but a slight hollowness to everything.

The Aperions weren't warm, but they were warmer than the Gurus. Rosie wasn't quite as far forward. The snare and cymbals were balanced with the bass drum and with the overall mix. But the definition wasn't as pronounced. There was no hollowness, but a slight warm blanket on everything.

The NHT/sub was midway between the Gurus and the Aperions. Deep bass, but lacking the great definition and texture of the Gurus.

The Gurus plus sub brought out the electric bass, but the midbass was a little muddy and some of the definition was lost.

Winner? The Gurus and the Aperions were two equally valid approaches. The Gurus really grabbed my attention with definition, but the Aperions might be easier to live with long term.

Liszt—Mephisto Waltz (Reference Recordings 25)—The Gurus are Goldilocks! Warm speakers enhance the bottom of the piano on this recording so that it sounds like a Bosendorfer (which it's not), but they also roll off the top end. Cool speakers make the piano sound thin on the bottom and clanky on top. The Gurus showed a full concert grand bass without reverb and a nicely extended but not at all clanky top end. Perfect—the most natural and enjoyable I ever recall hearing this recording.

The Aperions were much warmer. The top was rolled off, and the bottom had

added reverb and smear. It was still a natural concert feeling, but several rows back (the Gurus were actually on stage).

The NHT/sub had more bass, but not in the sense of deep bass extension. The hall was more reverberant, the mids felt warmly chesty, and the highs were rolled off. Overall balance was tipped far to the bass end.

The Gurus plus sub had no perception of added bass, but the mid-bass wasn't as clear.

The Gurus (sans sub) were the clear winner here, with a natural piano sound and wonderful presence.

Canteloube—Chants d'Auvergne (Virgin 90714)—The Gurus presented the sharply recorded voice very well. The edginess and buzzing that so often plague this recording were gone, but the voice didn't feel rolled off. The orchestra was well defined but didn't have the fullest bass extension. Warm speakers sound better on this recording, but the Gurus were probably as kind to it as you can be while still reflecting the actual recording.

The slight warmth of the Aperions was again a veil that went too far in taming this recording's problems and moved the whole performance several yards away. There was no vocal edginess, but the sound was much less involving.

The NHT/sub bass made the orchestra sound more balanced, but the warmth and rolled-off highs cut the involvement.

The Gurus plus sub were much like the NHT/sub.

Nobody wins on this recording (ever), but the Gurus were probably the best possible combination of warmth and accuracy.

Holst—Planets (EMI 47160)—On this recording and those that follow I always raise the volume to 82dB, and physics seemed to reassert its rules on the Gurus at that loudness. The sound was excellent from the cellos up and they were able to handle the loudest passages without congestion, but the deep bass was missing and the system approached the volume limit because blue warning lights started flashing on all the climaxes.

(I checked with Lars Erickson about this, and he reported that "the blue lights indicate that the speakers are approaching distortion levels unacceptable to designer Ingvar Öhman. As a rule, as long as the lights are lit less than half the time, the speakers are performing within the strict parameters that Ingvar specifies." So no harm, no foul.)

The Aperions were similar, but the highs were a little shrieky.

On the NHT/sub the bass drum and tympani were much more noticeable and natural, but the highs came on like a peaky turbocharger: wham!!! They had more peak headroom, but a very noticeable change in timbre as the tweeters took full power.

The subwoofer took the strain off the Gurus to keep the blue lights off and add some power to the bass drum, without any cost. This was the clear winner on this cut.

Prokofiev—Romeo & Juliet (Delos 3050)—But don't write off the Gurus for orchestral music without a subwoofer wingman. On this selection they once again got it just right, with full extension at both ends plus more detail and texture than I'm used to. The trombones felt perfect. This had a real "you are there" feeling.

The Aperions' slight warmth enhanced the trombones into a menacing gang (very appropriate for this Montague/Capulet rumble), but that was the only gain. There was less extension on top, no more depth to the bass, and much less definition and texture.

The NHT/sub was all highs and lows (when the highs are loud in the score). This was like a Detroit big-iron car that puts up good numbers but loses the comparison test because it just doesn't feel right. The trombones were really powerful, but they were surrounded by an unnatural orchestra.

The Gurus and sub didn't gain anything, because the added bass was clearly accompanied by mid-bass smear.

So the winner was the Guru pair, which provided all the extension plus detail and texture for a very "real" feeling.

Bach—Tocatta & Fugue (Delos 3503)—The Gurus were perfect again. The bass chords have all the extension and authority you need (from 4.25" drivers!). The treble pipes were fully extended but not edgy (this is a close-recorded CD, so that is often a problem). There was lots of detail and texture—I heard more notes in the chords than I've ever noted (pun intended). Hall reverb (which is tight on the recording) was reproduced just right.

The Aperions had typical good bookshelf performance, which shows how great the Gurus are. The feeling of spaciousness and just-right hall reverb were replaced by a slight warm muddiness in the middle, and the highs weren't fully extended and were slightly harsh. Bass extension was OK for 5.25" drivers—not really different from the Gurus.

The NHT/sub brought real bass to the party. Being able to reproduce the pedals at full volume counts for a lot on pipe organ music, but again this felt like all top and bottom.

The subwoofer's effect on the Gurus was subtle—a little more bass, a little congestion. Either way is OK, but for me the subwoofer doesn't cover the cost of admission. So I declare the Guru pair the winner again.

Bach—Tocatta & Fugue (Dorian 90134)—Speakers that do well on the Delos usually are too warm for the Dorian, but the Gurus handled the Delos without adding much warmth and they also tamed the excess warmth on this recording. The result still isn't satisfying, but I think the problem is in the recording—the hall is so reverberant and the mikes were so far away that the detail I long for just isn't there. What's notable is that the Gurus' sound is the best parlay for these two Bach recordings that I've heard.

The Aperions' added warmth on a warm recording only exacerbated the problems. Everything was submerged into the reverb soup. This could work if there were real bass extension, but with the satellites alone it was not at all involving.

Continued on p. 41.

Lipinski L-707 Loudspeaker

by James T. Frane



	Lipinski Sound Corporation
	6120 Massachusetts Ave. Bethesda, MD 20816
	www.Lipinskisound.com

The \$2,295 per pair Lipinski L-707 loudspeaker has red-stained, wood grain finish on all surfaces. It is the larger of their two speaker designs, both of which are two-way sealed enclosure designs. The L-707 measures 23.6" high x 9.4" wide x 12.4" deep and weighs 41 lb. The enclosures are made of 1" thick MDF (medium density fiberboard) with internal bracing to minimize unwanted resonance.

The loudspeakers are shipped very well protected in zipper carrying cases lined with polyethylene foam. These are, in turn, surrounded by foam beads in cardboard boxes. The carrying cases are an added benefit to those who move the speakers periodically, both for ease of carrying and for the superior protection they provide.

DESIGN AND PERFORMANCE

Three drivers are mounted vertically on the front baffle centerline of each loudspeaker. Two 7" glass fiber cone midrange-woofers are above and below a 1" neodymium ring radiator tweeter. The midrange drivers have rubber surrounds, die-cast chassis, and a magnet described as being low distortion.

The tweeter is said to have "super low distortion" and extra wide dispersion with frequency response up to 40kHz. It is recessed about 1½" so sound from the tweeter and midranges reaches the listener at the same time. The recess is surrounded by three stepped-back rectangular layers of Belgian foam. Lipinski states the foam is shaped for maximum accuracy of tweeter performance by minimizing edge diffraction.

Lipinski's website states that the lowest possible crossover frequency was selected to deliver the best transient response. Crossover parts are described as foil inductors wound on wooden cores, non-inductive resistors, premium audiophile-grade capacitors, and special flat, super OFC (oxygen-free copper) internal wires. Lipinski states: "Internal dampening [sic] is outsourced and precut to absolutely identical shapes and weights for the best speaker-to-speaker consistency. Lipinski speakers are all magnetically shielded for demanding pro video applications. Speaker performance was per-

separate cables in my system have not resulted in better sound.

The loudspeakers are designed to be placed on stands, cabinets, or shelves that put the tweeter at ear level for a seated listener. Stands are available on the Lipinski website, as are subwoofers to fill in the lowest octaves. Lipinski rates the overall speaker frequency response as 56Hz to 20kHz ± 1 dB (31Hz-40kHz ± 3 dB) with a sound pressure level of 90dB 1W/1m and impedance of 4 Ω . Their stated design approach with these speakers is to use a "sealed enclosure tuned for the best impulse response rather than strive for low-end extension." They believe vented enclosures do not have the same accuracy as their sealed enclosure design.

The speakers are offered with two grille choices. One covers the entire front baffle, while the other covers only the mid-woofers, leaving the tweeter exposed. You can select either grille design, although both came with the review samples. Or, you can leave the

"The Lipinskis are solidly constructed with apparent well-thought-out design and quality craftsmanship."

fectured using an anechoic chamber."

The back baffle has separate, five-way, gold-plated binding posts for tweeter and midrange woofers. Large nuts, each with three-finger-grip cutouts, are provided for easy tightening without tools. The design accommodates up to two gauge (AWG) wire, spade lugs, and banana plugs. The tweeter and midrange posts are connected with removable gold-plated straps so that you can use a single pair of speaker cables. I left the straps in place for this review, because past experiments running

grilles off entirely. Both grille types attach to posts on the baffle.

Lipinski has found that all grille cloths pose some restriction of high frequencies and states: "Our laboratory and listening tests prove that even the most transparent grill [sic] fabric causes high frequency comb filtering." Thus, depending on the décor, either use the grilles that expose the tweeter or remove the grilles for serious listening.

Because the tweeters are mounted between the midrange-woofers in a

D'Appolito configuration that controls vertical dispersion, the speakers must be on rather tall stands to put the tweeters at ear level for a seated listener. Given their 40+ lb weight, a sturdy perch is needed for stability.

The Lipinski website and the booklet furnished with the speakers makes the following statements: "The first step in proper audio reproduction is a well-designed room." They provide a PDF of a room depicted by their design team. Designing a room specifically for music or other entertainment system installation would be ideal, but many listeners may not be able to accommodate the details shown. Not to worry, because these speakers performed well in my more typical room. Lipinski then states: "Once you have a solid room, there are other rules that enable good sound reproduction" and proceeds to address matching components in a sound system, including amplification, cables, and using stands of the correct height. Lipinski designs and custom-manufactures 36" tall stands for their speakers.

I placed the speakers 9½' apart and each 9½' from the listening position. The Lipinski website shows their ideal audio-video room, and positioning info is included. Their site also includes a photo with the speakers on an ear height shelf facing a couch. Moving the Lipinskis to slightly different positions and orientations made some discernible differences; aiming them directly at the listening position was best. The closest sidewall was 5' and the distance to the wall behind the speakers was slightly over 4'.

LISTENING TESTS

I auditioned them with a wide variety of music and made a few comparison tests between them and my floor-standing Thiel CS 2.4 speakers using my ADC SS-525X 2/3-octave spectrum analyzer. The calibrated mike was at ear height at the listening position. Sources were pink noise and the Sheffield A2TB Test Disc. I found that from about 50Hz and up,

the Lipinskis were reasonably flat and were down a measured 8dB at 40Hz, dropping off below that. Because both drivers and enclosure are smaller than the Thiels, it was reasonable that the Lipinskis did not go as deep. You can compensate for this with a properly located subwoofer having an adjustable low cutoff frequency.

The Lipinskis reproduced sounds—including textures and timbres—accurately. Male and female voices came through with naturalness. Instruments sounded as they should. Details, depth, and spaciousness were all good.

The single greatest difference between the Lipinskis and my larger—and nearly twice as expensive—Thiels was the sound staging and depth. The Thiels were better in this area, projecting a wider and deeper soundstage. A typical example was Julie London's "Cry Me a River" from *The Best of Julie London* (Capitol/Rhino R2 70737). The stage width was from speaker to speaker with the Lipinskis and some sense of depth.

With the Thiels, Julie was at the center of a wide stage that extended beyond the speakers. Depth was greater and there was more of a three-dimensional presence. Instruments were spread behind her. Lipinski details and timbre were good and differences in recording and mixing quality were identifiable between examples of the same song/artist from different labels. Such differences show up only with very good speakers, demonstrating the quality of the Lipinski speakers.

Lester Young's "Sometimes I'm Happy" from *Lester Young Jazz Masters 30*

(Verve 314 521 859-2) sounded good on both sets of speakers, but I found the Thiels to have a little more spacious sound. Byron Janis' *Schumann Piano Concerto in A Minor, Op. 54* with the Minneapolis Symphony Orchestra (Mercury Living Presence 432 011-2) was a slight bit wider on the Thiels with a little more spaciousness. Another factor in these comparisons is that the Thiels and Lipinskis are different sizes and designs and could not occupy the same space. I positioned both sets of speakers the same distances from the wall behind them, and as close as possible to the same distances from sidewalls and the centered listening position.

The Lipinskis are different in many respects from my Thiels, but are nonetheless very good speakers. Living in "earthquake country," I prefer floor-standing speakers, but that is not necessarily a criterion for good sound. When purchasing any sound equipment, you should listen to equipment using music with which you are familiar and make comparisons at the same volume level. There is perhaps little that is as subjective as the sound of speakers. In addition, the listening room is an integral part of the sound of speakers.

The Lipinskis are solidly constructed with apparent well-thought-out design and quality craftsmanship. I believe they represent a good value. My wife and I found them to have more visual appeal with either of the two types of grilles in place compared with no grilles. Although both types of grilles have foam lining the inner edges to minimize edge diffraction (redirection of the sound), I thought the sound was slightly better with the grilles that did not cover the tweeters or no grilles, given a good recording.

Solid construction, good value, and excellent sound make for an attractive combination. You will need speaker stands or other means to elevate them off the floor to put the tweeters at the listener's ear height. If you are in the market for speakers in this price range, the Lipinski L-707s should be on your audition list. **rr**

Associated Equipment

- Bryston BP25 preamp and SST 4B power amplifier
- Thiel CS 2.4 speakers
- Thorens TD320 Mk III turntable with Grado Reference
- Platinum and Clear Audio Aurum Beta phono cartridges
- Grado PH-1 phono preamp
- Sunfire True Subwoofer Super Junior (not used for speaker comparisons)
- Sony CDP-XA-2ES CD player
- Magnum Dynalab MD100 FM tuner
- Kimber 8TC speaker cables
- Bryston balanced interconnect cables (preamp to amp)
- Vampire and WireWorld unbalanced (RCA) interconnects



Reliable Reviews

Continued from p. 38.

The NHT/sub proved my point above. The full and powerful bass gave the verb a reason for being, so that the experience was like being in a big Gothic cathedral. This is probably what Craig Dorian wanted this recording to sound like.

The subwoofer wasn't well integrated with the Guru (I had been warned not to try this at home; I hope the Norse god Thor doesn't spray rally-track mud in my eye for the transgression), so the added bass didn't really help the recording.

The NHT/sub combo was best on this recording.

Nowell—Riu, Riu, Chiu (Sampler 8606)—Wow. The Gurus had tremendous presence without any edginess. All these treble-focused instruments (flute, hammered dulcimer, tambourine) sound close up and natural. Hand claps and wood case raps sound completely natural.

The Aperions were different but equally valid. Not quite as forward, but a little mellower.

The NHT/sub was like the Aperions but even better, which is a tribute to the NHT sats because the subwoofer wasn't doing anything.

Thus, the Gurus and sub sounded exactly like the Gurus alone.

Hard to pick a winner here. Let's call it a tie between the Gurus and the NHTs.

LEDR and Depth Tests (Chesky JD37)—All four setups were similar on the Up, Over, Lateral, and Depth tests, except that the sound on the Gurus stayed well above the speakers, and the Aperions did an unusually good job of distinguishing between the furthest distances (12' versus 9').

CONCLUSION

I started out comparing the Guru QM10s to early Saab rally cars, which implies that they excel in a narrow category but wouldn't do as daily drivers. Yet, as I tote up the score, the Gurus won or tied on every selection but one. So what don't they do exceedingly well?

The blue light on the Holst recording suggests that they aren't the best choice

for heavy metal, the loss on the Dorian CD shows they can't do highly reverberant bass-heavy works with pipe organ or full orchestra, and the designer's correct admonition to avoid subwoofers means they won't fit in with a home theater. [Well, sending out such signals have not been my intention. On the contrary, they do work excellently in a home theater—if used with a good pair of subwoofers, or even better—the coming Guru bass modules made for them. If they are used together with a single, and less than superb subwoofer, they will still perform excellently, but the subwoofer will not.—Ingvar Öhman].

But on a wide range of two-channel music reproduction—from chamber music and jazz to tightly recorded full orchestral works—they are as unstoppable as the Stig [famous Swedish rally driver Lennart "Stig" Blomqvist, who was the inspiration for Top Gear's mysterious test driver—Eds.] in a Saab 99.

The only fly in the ointment is that the fall in the US dollar has caused a commensurate rise in the price of the Guru QM10s. At \$2,695 they are not cheap, and the optional piano lacquer finish is so worth it that you really need to figure \$2,995 (plus stands). The comparison speakers sell/sold for a fraction of that. But Dodge Darts sold for a fraction of Saab 99s, and I know a lot of Saab fanatics compared to zero Dodge Dart lovers.

So the Guru QM10s are not for everyone, but if my description of their strengths fits your musical taste, I highly recommended them.

Additional note from the manufacturer: As Mr. Koster writes in the review, we recommend avoiding subwoofers in the standard setup (one woofer). However, we are currently at preproduction stage of our own "base module" to be used with the QM10. We recommend using them in pairs and it is intended to replace the QM10 stand.

Erik Ring
Guru Pro Audio

rr



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READERS REACT

I read the latest (Sept.) *audioXpress* and it is apparent that times are tough for a lot of industries. I took your idea of getting new subscribers and subscribed for a friend of mine as a gift. If every one of us who subscribes could do the same thing, I imagine it would help *audioXpress* a lot.

The second thing that I am going to do is not accept money for any submission I make. I realize this is a small amount of money, but every little bit helps. I do not want *audioXpress* to fold!

Bruce Brown
TuninFork@aol.com

You can tell your advertisers I went right out and bought a Sony XDR-F1HD radio tuner after reading the review in the August issue and checking K6sti's website. I doubt it will resurrect WQXR (NYC classical station) after it switches over to whatever dismal fate is ahead, but it'll give me a fighting chance with the remaining suspects.

J.G. Owen
Huntington Station, NY

I have recently begun using one of your advertisers as a supplier—PCBCart. The first occasion was boards for Stuart Yaniger's Impasse preamplifier (Feb. '09) which were distributed via DIYAUDIO as a group buy. Just yesterday I sent along another order for power supplies boards for the same. I was pleasantly surprised with the quality.

Jack Walton
jdwalton@comcast.net

After reading the editorial in *aX* Sept. 2009, I was very unpleasantly surprised with the reality you described. Could it

be true that the bloody crisis could eject the only audio magazine existing now? Maybe that means that the Asian electronic factories won the war, by surviving, and all the rest of the manufacturers will disappear?

I express my support and my solidarity in all your efforts to give us the best news, and for my part, I'll try to help as much as I can. I will renew my subscription for two more years. And I will find some new subscribers here in Greece and in Poland or in Romania. Anyway, if this year things do not get better, I intend to close my little laboratory, too. I shall send some prototype amps to my son in Poland, make some presents to my friends, and continue to write my memories. Sad but true!

Alex Arion
Athens, Greece

SOUND DIFFERENCES

I read and enjoyed the editorial in the July issue, "Amplifier Differences and Lack of Tests." I will pass along some recollections I have going back to the 1940s that may be of interest to other hobbyists. I agree that listening tests should not be the sole criterion in judging equipment, particularly the electronics, despite the fact that listening is what we finally do.

Going back to the vacuum tube days, audio buffs often said triodes sounded different from pentodes as power amplifiers. They usually did. The pentode characteristically has a very high AC plate resistance compared to that of the triode. This accounts for most of the difference because they were usually employed (without feedback).

Without feedback, there is a noticeable difference in speaker damping and significant variation in output frequen-

cy response due to speaker impedance variations versus frequency, especially pronounced with pentodes. Feedback largely corrects these problems when properly applied. Slightly greater power output can usually be obtained from pentodes rather than triodes of the same plate dissipation rating.

Single-ended versus push-pull. When the tubes are balanced, either triode or pentode, push-pull operation will result in waveform symmetry, thus eliminating even-order harmonic distortion. Greater power output also occurs with push-pull operation due to two tubes delivering power and a little due to the waveform improvement itself. The push-pull configuration allows a more power-efficient class of operation—AB or B—than A (A is often preferred, however). Feedback provides the same benefits in push-pull as in single-ended operation.

Transistors versus tubes. Transistors share some qualities with tubes. Bipolar transistor collector current, analogous to tube plate current, is controlled by its base current, analogous to the tube grid voltage. Field-effect transistor drain current, analogous to tube plate current, is controlled by its gate voltage, analogous to the tube grid voltage. The bipolar transistor is current controlled, whereas the field-effect transistor is voltage controlled, as are vacuum tubes. Both transistor types have characteristics resembling the pentode, in that the collector or drain current is nearly independent of voltage and its current is controlled by the base or gate.

Some hobbyists believe that their systems sound better without feedback and also that single-ended sounds better than push-pull. I agree that such a system would sound *different*, but *better* is

a subjective term. No doubt there would be significant distortion, including even-order, and the frequency response somewhat ragged.

My listening experiment using a function generator as the signal source is that a pure sine wave has pitch but no timbre and is not particularly pleasant; a pulse also has pitch but can be pleasant depending on pulse shape. A pulse is rich in harmonics. This raises the question: Is a little distortion a good thing? To these hobbyists it must be.

In the design of all professional audio equipment, and most current quality consumer gear, it is evident that low distortion, flat frequency response, and low noise are paramount. Since the 1950s, LP records have been mastered with feedback cutters, pioneered by the Westrex cutter. The advent of the elliptical playback stylus made a contribution, too. As far as I know, all other forms of recording material—from microphone to the recorder—have employed feedback electronics.

In the schematics I have seen of 1930s and 1940s radios, I do not recall any of them having feedback, although feedback was then known, but I believe was mostly employed in the telephone system. Many of the radios had pentode output stages. Tabletop radios usually had single-ended outputs, and most of the consoles had push-pull outputs, some with triodes. The radios always seemed to be deficient in the higher audio frequencies. I recall some people saying such radios had a “mellow sound” which they liked.

I don't think it matters whether tubes or transistors are used as regards listening quality if the design is well done. Transistors have an edge as regards cost; they generally cost less for a given dissipation rating, and output transformers can be avoided, for a couple of reasons. To our friends who might like a little distortion, this could be arranged by inserting something in the signal chain having a square law transfer characteristic to provide even-order harmonics.

*J.L. Markwalter, Jr.
Port Charlotte, Fla.*

I would like to respond to the interesting editorial by Kent Smith. I believe I

have a rational explanation for the differences in sound that he noted between tube and transistor amplifiers.

One difference is due to different frequency responses. Both amplifiers may have a flat frequency response when measured with a resistive load, but loudspeakers are seldom resistive. With a loudspeaker there will be a different frequency response with an amplifier that has a high source impedance (tube amplifier) as compared to an amplifier with a low source impedance (transistor amplifier). The tube amplifier with a high source impedance will have more output when the loudspeaker impedance is high than when it is low.

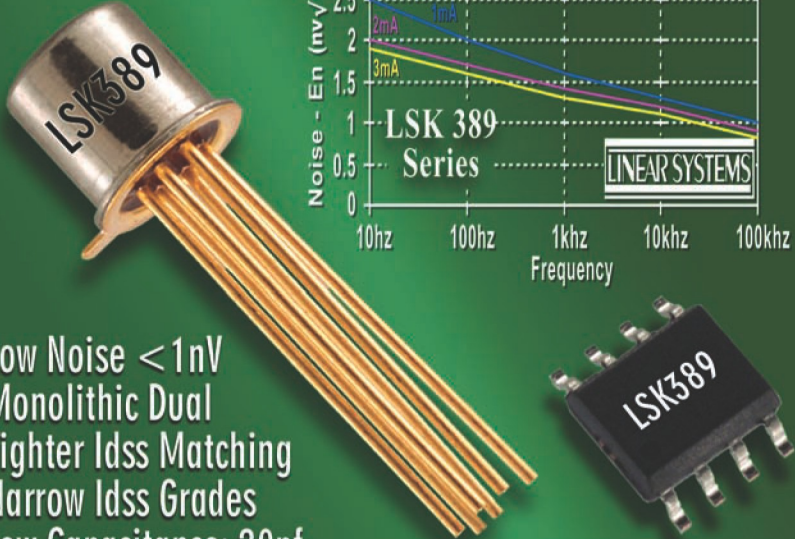
Consider the woofer in a two-way loudspeaker. At low frequencies (60–120Hz) the woofer has a resonance that makes its impedance high, and this will result in increased output with the tube amplifier. At higher frequencies (near crossover), the woofer will have increased impedance due to the woofer voice coil inductance, again resulting in increased output from the tube am-

plifier. These increases in output will vary depending upon the source impedance, but can be as high as 6dB. Such increases are easily heard, and may be interpreted as “cleaner and more detailed.”

This is not an unknown phenomenon. John Atkinson (of *Stereophile* magazine) comments on the impedance of a loudspeaker in the June 2009 issue (p. 60), “The impedance is significantly higher in the region covered by the latter two drivers. . . which means that the speaker will sound tilted-up at higher frequencies when used with a tube amplifier having a high source impedance.”

An easy way to test for this phenomenon is to put an equalizer before the tube amplifier and equalize the frequency response of the tube amplifier (with the loudspeaker connected) to be the same as the transistor amplifier. It will be difficult to achieve perfect equalization, but variations of 1dB are OK. Then listen to the tube and transistor amplifiers and see whether the differences have largely disappeared. In my

1nV Low Noise Dual JFET



The image shows a cylindrical metal package of the LSK 389 Dual JFET with four gold wire bonds extending from it. To the right is a smaller, black plastic package of the same device. Above the packages is a graph showing Noise - En (nv/√hz) on the y-axis (0 to 3) versus Frequency on the x-axis (10hz to 100khz). Three curves are plotted for bias currents of 1mA (blue), 2mA (purple), and 3mA (yellow). All curves show a decreasing trend in noise as frequency increases. The 1mA curve starts at approximately 2.5 nv/√hz at 10hz and drops to about 1.2 at 100khz. The 2mA curve starts at approximately 2.0 and drops to about 1.0. The 3mA curve starts at approximately 1.8 and drops to about 0.8. The text 'LSK 389 Series' and 'LINEAR SYSTEMS' are visible on the graph.

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experience, the frequency response differences are more readily heard than other causes.

This is not to say that other causes do not exist. But first, try to get the frequency responses of the amplifiers the same.

*Dick Crawford
Germantown, Md.*

Kent Smith responds:

I think Mr. Crawford's suggestions are great. I am glad that someone has tried to use EQ to level the field and observe the results. I am a little mystified about the results he mentions, because none of the tube amps I have built give more bass than a transistor amp; in fact, they all sound as though they are putting out less bass.

I have tried SE triode, PP triode, and PP pentode UL with similar results. I do understand the technical reason why they should put out more into a high Z. I do know that transformers have more reactance at their frequency extremes, and this could combine to affect response—yet another subject I have not seen any information on! I don't have any EQ equipment, so I hope someone else can verify this and send the results to *audioXpress*.

I should mention that I don't have any "audiophile" transistor amps, nor have I built any of Nelson Pass' designs, though I plan to. I recognize that designing for low distortion before adding a little feedback could produce much better results. Along those lines, I designed a little test.

I had ordered some MOSFETs for an amplifier that I never got around to, and discovered some PA/speaker transformers that had a low enough turns ratio to match the output of a single MOSFET. This allowed me to build the same Class A SE transformer coupled amplifier in both MOSFET and triode. Because the frequency response of the transformers were not far off, I expected pretty similar sound, which was the case.

But after a minute, some small differences started to stand out. The overall balance was the same, but the MOSFET just wasn't quite as lively. Of course, this is very subjective, and I prefer the sound of the triode, but I don't know which is more accurate.

While I won't say from this that a MOSFET can't sound as good as a triode; one way or another, it probably can (maybe as a source follower?). It just looks as though it's easier to get the best results from a triode, at least for a home builder like me with minimal equipment. I would not be surprised to hear now that some of the best audiophile amps are as good as tube amps. I just can't afford them!

But my question remains, where is the data? Have we proven what distortions (and levels) are audible, and therefore, important? Do we know what happens to the response and distortion levels when a reactive load is hooked up to highly damped amp, an impedance matched amp, or a transformer?

Hearing other people's impressions is nice, but seeing good data and proven explanations is even better. Anyone game?

Having written many articles about my experiences with circuit design, sound recording, as a band musician, and doing tons of listening, I found Kent Smith's editorial fascinating and appropriate to the need for new amplifier testing methods. First, I'd like to mention my relevant *aX* articles:

1. Intrinsic Fidelity Testing (Nov.'07).
2. An AB/I Switch (June '07).
3. Mad Katy (June/July '06)—a purist 250W tube amp.

The Intrinsic Fidelity Test (direct, precise level-matched sonic comparison of an electronic component's input versus output signals, *real-world loaded*) is extremely sensitive *and* removes taste preferences, and source quality degradation (be it good or bad sounding) is listened to.

With my Mad Katy amp, reactively loaded, I could hear *no* input/output difference (and my hearing has recently been tested to be very good). This agrees with my live versus reproduced voice and bell tests with Mad Katy and solid-state amps—the sounds were very natural with the tubes, but somehow "constricted" and "electronic" with solid-state.

Somewhere in *aX* I once suggested using a large number (9 or more) of log-spaced test sine tones for a stimulus, *and*

(as you said) a real-world reactive load, with power amp testing.

My experiences with clean acoustic music agree completely with yours. One comment I have is about your suggestion of an over-damped response: First, it's an *under*-damped response that prolongs tones. But more important, a power amp can't be "over damped." An *infinite* damping factor (DF) simply means zero output impedance, which would drive the speaker with a perfect frequency and transient waveform signal (constant voltage), assuming all else was perfect in the amp.

Most speakers (the good ones, at least) are designed for constant-voltage (versus frequency) drive. Of course, a high amp output impedance (such as from a SET) might *sound* good because its high Zout makes the speaker's input response follow the speaker's impedance—this adds a narrow boost at the speaker's resonant bass impedance rise, plus other variations that just *might* perceptually "liven up" a dry, compressed recording.

But I digress—your editorial calls for a real need, I believe. Incidentally, regarding your perception of sub-1% higher harmonics, audio researchers have long established that the hearing threshold of harmonics becomes more and more sensitive as the order number is increased regarding the 2nd harmonic. Now factor in the *thousands* of spectral components in real music—fundamentals, harmonic and other overtones, transient spectra (a dense continuum) plus modulation sidebands (tremolo and vibrato)—trying to make that poor amp follow this into a reactive load—and it's no wonder that our present tests are far too primitive to correlate with sound!

*Dennis Colin
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aX*

SHOW REMINDER

The third international Burning Amplifier Festival will be held October 18 at the Presidio Yacht Club in San Francisco/Sausalito. For more, visit www.burningamp.com.

The Munich High-End Show 2009

By Ward Maas

Crisis? What crisis?

The CES in Las Vegas at the beginning of the year showed a significant decline in the number of visitors and exhibitors. The media, constantly looking for any indicators of a global economic crisis, saw this as an omen. Figures from the Munich High-End Show, however, reflect an improving trend—248 exhibitors (230 in the previous year) from 26 countries.

The show featured the introduction of many new products. There were many re-polished current products with a “new” sticker, but also a few products that showed that some development departments have been rather busy.



PHOTO 1: High-End logo.

PRODUCTS ON DISPLAY

This audio fair hosted a sea of expensive amplifiers, loudspeakers, and other related gear. It seems that only large, huge, and ridiculously huge are the keywords when it comes to quality audio reproduction, as manufacturers vie for the number one crown. For example, the Toruspower (www.toruspower.com) mains conditioner, with a Plitron toroidal transformer, beat everything I have seen for the last years (Photo 2). At the Angstrom research booth, a grand-piano-shaped turntable with a turntable arm made of a violin bow (Photo 3).

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“The Munich High-End Show 2009,” by Ward Maas.

Measurement tips for Mark Williamsen's Dipole Microphone (aX 6/09).

“Rebuilding a Cheap Chinese EL84 Amp,” by Merlin Blencowe (aX 7/09).

Circuit Boards from Tom Perazella's “Maintaining Your Balance, Part 2” (aX 8/09).

Construction Guide from Jan Didden's PaX, Part 2 (aX 9/09).

Articles from Past Issues:

NEW! Reliable Reviews:

Pioneer X-Z9 Compact Stereo System, Pt. 1, By David A. Rich (aX 9/09).

Echo Busters Decorative Acoustical Treatments, By Tom Lyle (aX 9/09).

THIEL SCS4 Loudspeakers, By James T. Frane (aX 8/09).

Grado iGrado and SR-80i Headphones, By Tom Lyle (aX 8/09).

Sony XDR-F1HD, By David A. Rich (aX 8/09).

“Design and Measurement of a Dipole Microphone,” by Mark Williamsen (aX 6/09).

Review of *The Guitar Player Repair Guide*, by Chuck Hansen (aX 5/09).

“Building the Occam Audio MTM 4 Kit,” by Bill Fitzmaurice (aX 3/09).

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