**Showcase: A 70W McIntosh Amp Rebuild**

# audioXPRESS

THE AUDIO TECHNOLOGY AUTHORITY

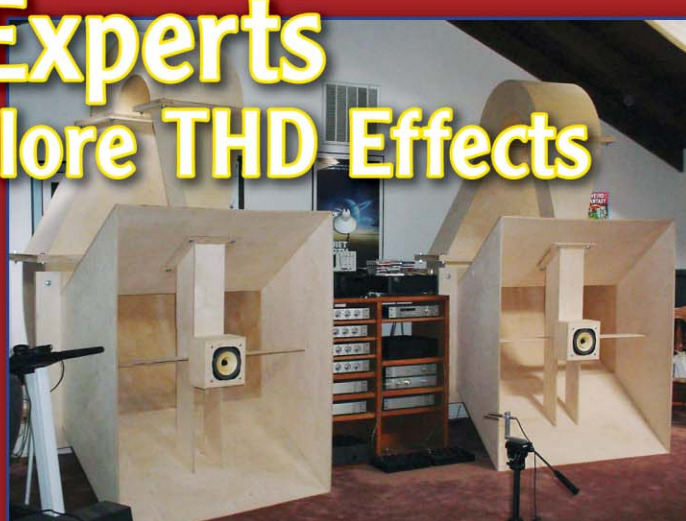

OCTOBER 2004
US $7.00 Canada $10.00


Celebrating 35 Years

## Adventures with Full-Range Efficient Drivers

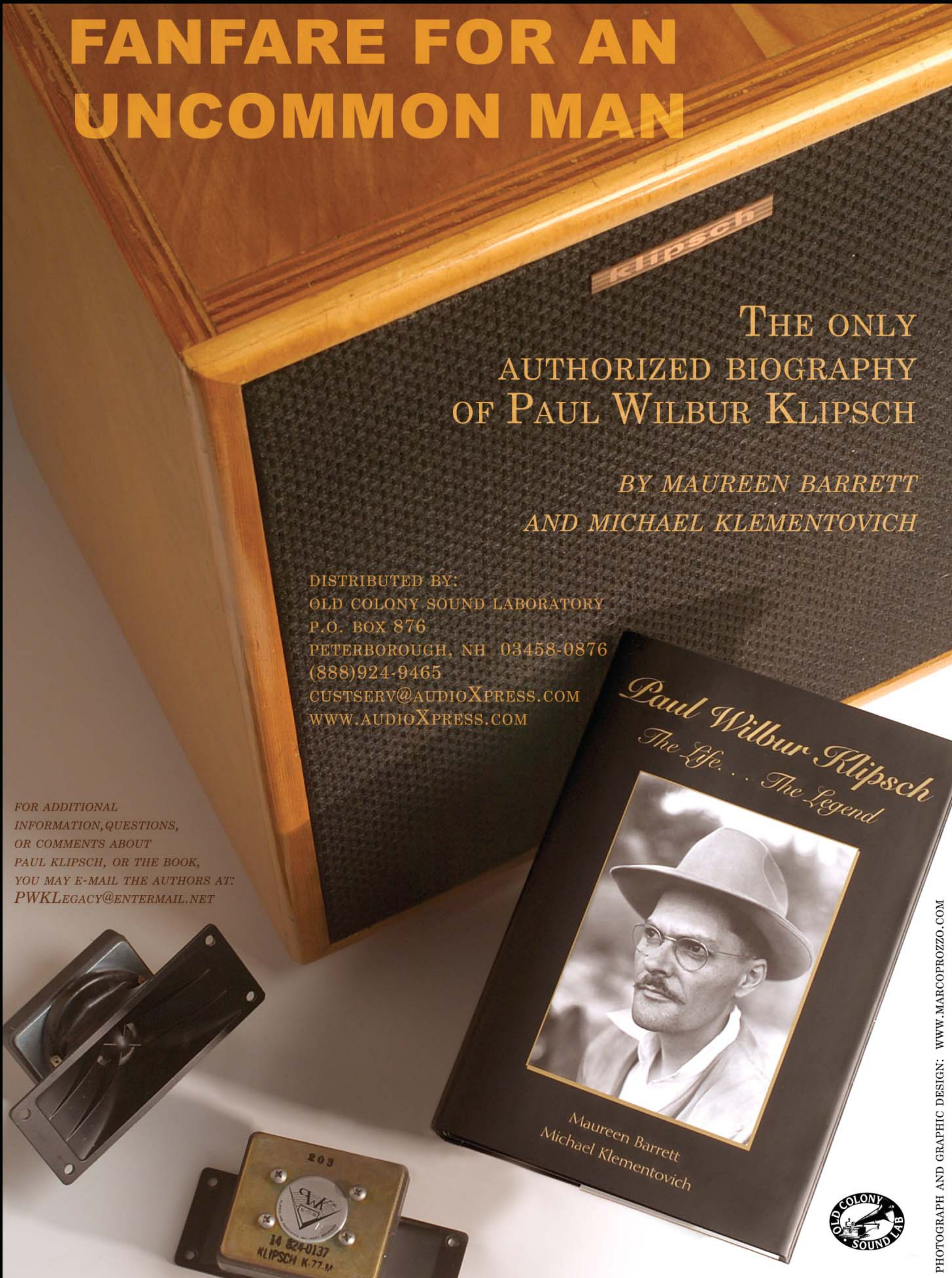
## Build a Unique Tube Control Unit

## 3 Experts Explore THD Effects



## Preview: Speaker Building 201


www.audioXpress.com
10
0 74470 84697 0




PHOTOGRAPH AND GRAPHIC DESIGN: WWW.MARCOPROZZO.COM audioXPRESS

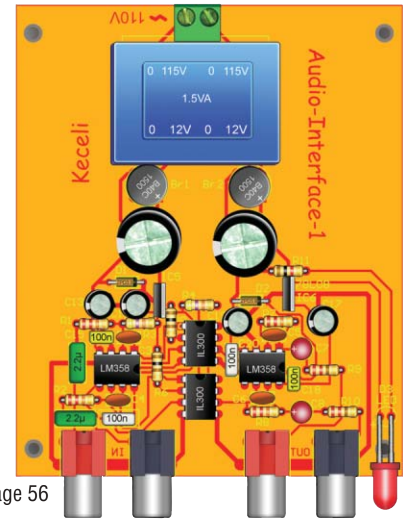
# CONTENTS

VOLUME 35 NUMBER 10 OCTOBER 2004

## FEATURES



Page 26




Page 56

## DEPARTMENTS



Page 6

## IN EVERY ISSUE






*audioXpress* (US ISSN 0004-7546) is published monthly, at $34.95 per year, $58.95 for two years. Canada add $12 per year; overseas rates $59.95 per year, $110 for two years; by Audio Amateur Inc., Edward T. Dell, Jr., President, at 305 Union St., PO Box 876, Peterborough, NH 03458-0876. Periodicals postage paid at Peterborough, NH, and additional mailing offices.

POSTMASTER: Send address changes to: *audioXpress*, PO Box 876, Peterborough, NH 03458-0876.





**SUBSCRIPTION/CUSTOMER SERVICE INQUIRIES**
A one-year subscription costs $34.95; a two year subscription costs $58.95. Canada please add $12 per year. Overseas rate is $59.95 for one year; $110 for two years. All subscriptions begin with the current issue. To subscribe, renew or change address write to the Customer Service Department (PO Box 876, Peterborough, NH 03458-0876) or telephone toll-free (US/Canada only) 888-924-9465 or (603) 924-9464 or FAX (603) 924-9467. **E-mail: custserv@audioXpress.com. Or online at www.audioXpress.com**
For gift subscriptions please include gift recipient's name and your own, with remittance. A gift card will be sent.

**EDITORIAL INQUIRIES**
Send editorial correspondence and manuscripts to *audioXpress*, Editorial Dept., PO Box 876, Peterborough, NH 03458-0876. **E-mail: editorial@audioXpress.com.** No responsibility is assumed for unsolicited manuscripts. Include a self-addressed envelope with return postage. The staff ***will not*** answer technical queries by telephone.

**CLASSIFIEDS & WEB LISTINGS**
Contact Nancy Vernazzaro, Advertising Department, *audioXpress*, PO Box 876, Peterborough, NH 03458, 603-924-7292, FAX 603-924-6230, **E-mail nancy@audioXpress.com.**

# nextgen™

## The exact solution for all digital connections


- the perfect match for all digital audio and video links (75Ω)
- true wideband RCA type connectors
- free from interferences
- with pure copper or silver contacts
- ultra rapid signal transfer

Ph: 801-621-1500
www.wbtusa.com

WBT

# Current Source Amps and Sensitive/Full-Range Drivers

This survey article investigates loudspeaker performance when driven by a current source amp.

**By Nelson Pass**

PHOTO 1: Lowther DX55 in the KleinHorns.

Conventional wisdom holds that a pure voltage source amplifier is ideal for audio applications, and generally designers of loudspeakers work to that assumption. This belief has particularly been dominant since the development of high power solid-state amplifiers in the 1960s. A small minority of audiophiles thinks otherwise, and these people often use low wattage tube amplifiers with unusual-looking speakers. Well, of course entertainment is full of fringe elements.

A couple of years ago Kent English and I were playing around with various ribbon tweeters, noting that without a matching transformer the ribbons themselves seemed perfectly happy driven by a current source. We built an audio current source that delivered 1A of AC current per input volt to a maximum of 10A. This kind of amplifier is known as a power transconductance amplifier.

Such circuits are fairly common as small chips, but those do not have the high current needed to drive a loudspeaker. The amplifier had a high output impedance, and thus no damping factor, but we found that the ribbon in the tweeter didn't seem to need a damping factor, measuring and sounding a bit better without the matching transformer. We decided that a current source was likely delivering more accurate force/acceleration to the ribbon than a voltage source, and after playing with it for a couple of days we put it away and went on to other projects.

Also about this time Kent and I began playing with full-range drivers and loudspeakers that deliver bass, midrange, and treble with one cone. In a number of ways they don't measure as well as multiple specialized drivers, particularly at the bottom and the top, but there is something aesthetically appealing about the simplicity of the idea, and on many occasions they manage to sound very good, especially driven by tube amplifiers.

Most interesting are the full-range high-efficiency drivers that deliver the goods with only a watt or so. It's a big challenge to a designer to deliver a good-sounding full-range acoustic transducer with 100dB/watt efficiency. When it is well achieved, you get a wealth of detail, exceptional dynamic range, and a sense of musical "live-ness" that you don't often hear elsewhere.

Tube amplifiers seem to bring out the best in such drivers. They have more bottom end, a warmer, mellower mid and upper mid-range, and often more top octave. By comparison, the "best" solid-state amplifiers make them sound more like transistor radios—less bottom and an occasionally strident upper midrange. If you are a solid-state kind of guy (like me), you start wondering how that could be, and if you are a tube aficionado, you smirk and say, "I told you so." The solid-state guy probably starts fixing the response with a parametric equalizer, and the tube guy enjoys his music with a nice glass of wine.

Critical damping—that resistive combination of electrical source impedance, suspension friction, and acoustic

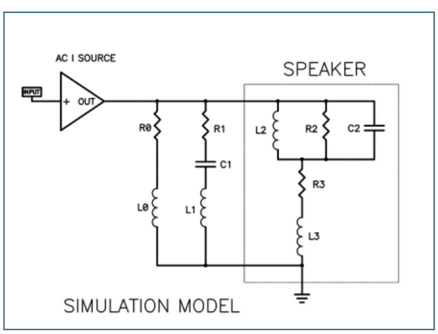



FIGURE 1: Speaker model. B-2466-1

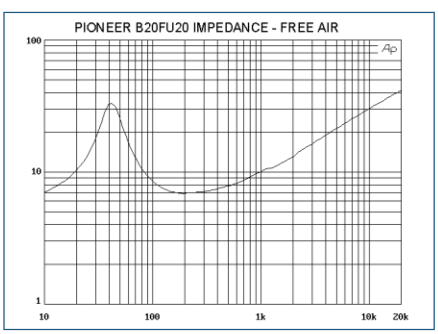



FIGURE 2: Pioneer impedance. B-2466-2

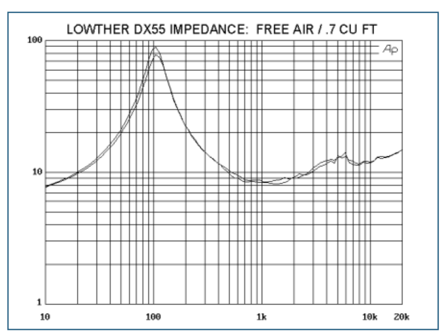



FIGURE 3: Lowther impedance. B-2466-3


Just like you would make.
Clean, efficient designs, manufactured from ultra pure Copper, non magnetic eutectic Brass, solid Silver, Teflon®, glass filled nylon, Rhodium, Gold and Silver plating... stuff that really works.
cardas.com
Component Parts

load—occurs when you apply a step pulse to the voice coil and the cone's motion doesn't overshoot. Under-damping results in bass notes that hang around a little longer than the amplifier intended. Over-damping has good bass transient control, but also significant loss of bottom end response. Generally we want something in between—something near to critical damping, and whether we under-damp or over-damp slightly seems to be a matter of taste.

The need for electrical damping is different for each type of loudspeaker and acoustic environment. High-efficiency full-range drivers are more easily damped than other types due to powerful, efficient motors and light cones. Looking at their bass response curves, we conclude that they are easily overdamped, resulting in excessive loss of bottom end. This partly explains the preference for tube amps with such loudspeakers.

Anyway, this assortment of observations arrived at a confluence when I hooked up a Son of Zen amplifier (*Audio Electronics* 2/97) to a pair of Fostex 208Es in sealed enclosures. The

PHOTO 2: Jordan J92S.

Son of Zen operates without feedback and has an output impedance of about 16Ω, which would be a damping factor of 0.5—minuscule compared to the 100 to 1000 you can get with regular solid-state amplifiers.

With the low damping factor it was a totally different speaker. It suddenly had bottom end response and a better top end. It still had the same annoying upper-midrange that made Dick Olsher (www.blackdahlia. com) devise his passive equalization network. The low damping factor didn't cure the upper-midrange faults, but it seemed to work improvements everywhere else.

A year later, Kent has acquired about 20 different full-range/mid- and high-efficiency drivers for us to play with (it's part of his job description). We've spent as much time exploring them as we reasonably can, trying different things to coax the best sound out of them with a current source amplifier and various passive parallel networks.

## WHAT IS A HIGH-EFFICENCY/FULL-RANGE LOUDSPEAKER?

To get high-efficiency and wide bandwidth, we fundamentally need two things—a great motor and a great radiating surface. A great motor means getting the most force/acceleration for the least amount of electricity. It means a big magnet with lots of magnetic density in a precisely machined gap where a very light voice coil sits (precisely). This voice coil is wound in a cylindrical assembly that maximizes the current exposed to the magnetic field and generates the most amount of force (acceleration) for a given amount of electrical current—in other words, an expensive motor.

The other half of the equation is the cone assembly attached to the voice coil. Here we start seeing more of the art that is involved in such a speaker, in contrast to the fine motor engineering. The radiating surface for such a loudspeaker must be very light, to maximize the acceleration from the voice coil. At the same time, the cone needs to be stiff and inflexible, so that this acceleration can be accurately transmitted to the entire radiating surface at once.

When this fails at high frequencies (which it will), the cone needs to decouple the force gracefully to a smaller and smaller surface so that it becomes effectively smaller at the highest frequencies. In practice it's a lot of art coupled with even more trial and error. Thomas Edison could have made a fantastic version of such a speaker, and for his time I believe he did.

Besides obviously requiring low power and only one driver per channel, the advantages of a full-range efficient loudspeaker are found in three characteristics. First, the speaker requires no crossover network to apportion frequency bands to different drivers, and does not suffer the phase shifts that come with such filters. Second, the sound radiates from one source, so that diffraction effects between drivers are removed. Third, the electrical and mechanical qualities required by such drivers give rise to subjectively good dynamic range and detail.

This is assuming an ideal example of a full-range efficient driver. The reality can fall short of this promise. Real drivers tend to be too limited to satisfactorily deliver both the top and bottom octaves of audio. The decoupling of high frequencies on the cone is often not smooth, and with that comes response peaks and dips in the upper midrange, often followed by a high-frequency response rolloff.

On the bottom end, over-damping of

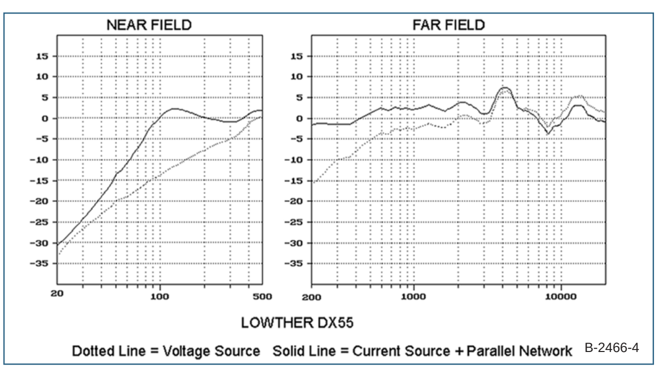



FIGURE 4: Lowther near and far field response.

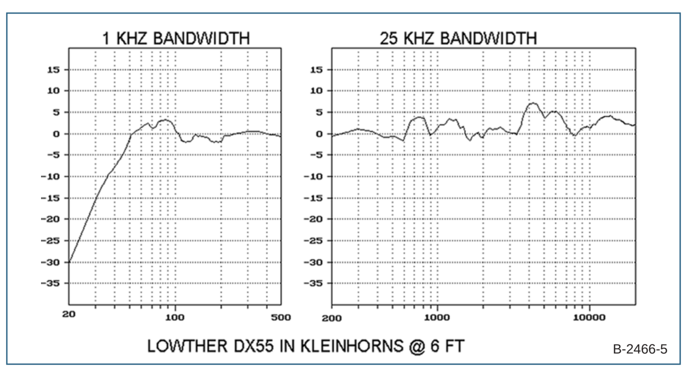



FIGURE 5: Lowther in KleinHorns near and far field response.


Hi-Vi RESEARCH®
hear different.
Hi-End OEM drivers for Professional, Home Theater, Multimedia, Car, Inwall/Ceiling, Subwoofer......
Leading on the Chinese market in Hi-end drivers and speaker systems production.
Produced over 3 million Hi-end drivers annually.
Produced over 600,000 Hi-end speaker systems annually.
ISO 9002 certified for all our drivers and speaker systems.
Two separate facilities that total 1 million sq. ft are dedicated to driver and cabinet production.
We are the first manufacturer to use LMS / LEAP in China since 1993.
Our equipment utilizes the best testing systems - B&K, TEF, MIssa, Clio and LMS to ensure perfect quality.
We also use advanced software including Ansoft, Redrock, Protel, 3D Max and Solidworks for cutting edge design.
There are over six hundred dealers and over 200 our own retail shops within China.
Our products undergo rigorous design and innovative engineering by professionals in the US and manufactured in China.
SWANS SPEAKER SYSTEMS, INC. 2550 Corporate Place, Suite C-103, Monterey Park, CA 91754 USA
Tel: (323)-881-0606 Fax: (323)-881-0581 e-mail: sales@dulcet.com
web: www.swanspeaker.com SWANS SPEAKER a division of Hi-Vi RESEARCH®

the very light cone contributes to a rolloff starting at an octave or more above resonance, so that many examples of such drivers fall short below 100Hz—in a closed box with a voltage source their response is as much as 15dB down at resonance. Some of these drivers are better than others, most of them are delicate, and some of them are very expensive.

## POWER TRANSCONDUCTANCE AMPLIFIERS

This is a fancy name for a power current source. An input voltage causes the amplifier to deliver a proportional output current. Of course, this sort of thing would occur with an ordinary amplifier driving a pure resistance, but a loudspeaker circuit is not pure resistance. It possesses numerous reactive elements, some due to inductance and capacitance in the electrical circuit, and some from the reaction to motion of the voice coil in a complex mechanical system. Fed by a voltage amplifier, the current through the loudspeaker voice coil is not directly or instantly proportional to the input to the amplifier.

Ordinarily, loudspeakers are designed around this assumption, but the "piston model" of loudspeakers presumes that over a specific range the acoustic output mirrors the acceleration of the voice coil/cone assembly, and this is reflected by the current through the voice coil.

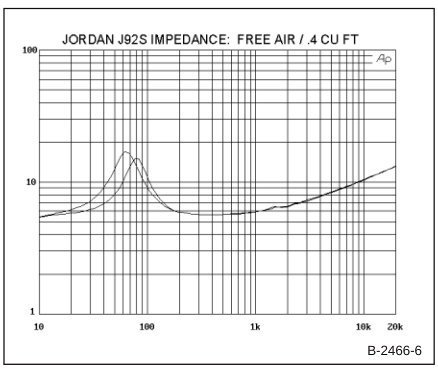



**FIGURE 6: Jordan impedance.**

The most precise way to develop that specific current is with a current source amplifier. Such an amplifier ignores the impedances in series with the circuit, the resistance and inductance of the wire and voice coil and the back electromotive force produced by the cone motion. As I said, most speakers are designed around voltage sources, but there are few instances where a current source can be used to advantage. One of the best ones is the category of full-range high-efficiency drivers.

Why is that? First, such drivers are able to take advantage of acoustic and suspension resistance to achieve some or all of the damping that they need to prevent excessive overhang because their moving mass is very light. With their efficient motors, even a high source impedance is often enough to give critical damping.

Second, their impedance curve tends to reflect their needs—more current at the low-frequency resonance and at the high end, both areas where the output response has fallen off with increased speaker impedance. If you want, the current through the voice coil can be made constant regardless of the variations in the acoustic environment. The voice coil force is invariant whether the cone is loaded into a horn, sealed box, bass reflex, or whatever else you care to mount it in.

Third, they are very easy to adjust in the upper midrange and high-end where you typically run into peaking problems. Most of these drivers are too hot in the upper midrange, and quite a few of them fail to make it to 20kHz. With a simple parallel network, you can arrange for a midrange dip followed by an increase at the top, evening the speaker response. More often on the higher quality drivers, you simply want an equalized "shelf" where everything above a specific frequency is evenly attenuated.

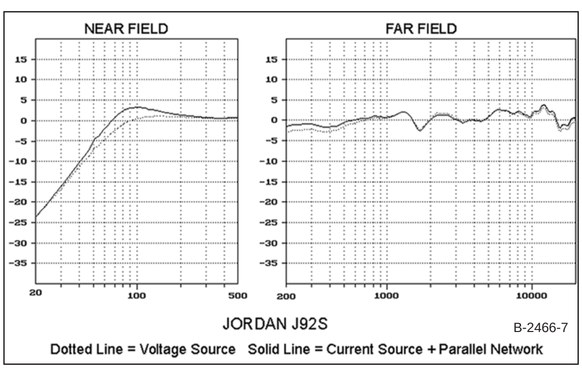



**FIGURE 7: Jordan near and far field response.**

The adjustments for damping at the low end and the mid/high peaking is easily accomplished using R, RC, or RLC (resistor/inductor/capacitor) networks wired in parallel with the output of the amp. You occasionally see this sort of equalization with full-range high-efficiency drivers with voltage source amplifiers, but there the elements are paralleled and then placed in series with the driver. With a current source, the equivalent circuit is the elements placed in series and then paralleled with the driver.

Of course, there's the added benefit that the resistive element of the speaker cable and the connection points can be made to largely disappear with a current source. To obtain the best effect, the loading network is placed close to the driver instead of the amplifier.

The following pages will show what we found with a variety of full-range sensitive loudspeaker drivers when we drove them with an active current source and played with parallel networks to shape the response to our liking. Most of the examples we will examine do not require true current source amplifiers, only amplifiers of quite high output impedances. Most of these cases will be happy with approximately 47Ω or so output impedance, and prefer 47Ω loaded in parallel with the output of a current source. That being the case, you can build a Thevenin equivalent of such a current source by placing a large resistor (later here to be known as R0) in series with the output of about a high wattage voltage source amplifier, and get something similar.

I'm not saying it will equal a spiffy First Watt F1 (being Class A and no feedback and all), and your resistor will run hot. On the other hand, you probably already have such a voltage source amplifier, and some of these speakers are quite cheap, allowing you a taste of these forbidden pleasures without high expense.

I want to emphasize that this does not serve as any sort of comprehensive guide to designing systems around this approach. You can build up the examples, and they will probably measure about the same, but remember that mostly we are restricting ourselves to the sealed box case for simplicity, and most of the time in the real world they are used in an enclosure that also uses the back wave of the driver. Also, the best measured curve is often not the best-sounding one, so be prepared to try various values to get satisfactory sound.

## AMPLIFIER/LOUDSPEAKER/EQ NETWORK MODEL

*Figure 1* shows a current source amplifier, the simplified model of the loudspeaker's impedance, and the generalized network that we can parallel with the speaker to enhance its performance. The speaker model consists of L2/R2/C2 that model the fundamental resonance, and R3/L3 that give us its DC resistance and high-frequency inductance. A good example of this is the simulation of the Pioneer b20fu20–51 driver. The values L2 = 0.05H, R2 = 27Ω, C2 = 300µF, R3 = 7Ω, and L3 = 0.1mH give a reasonable facsimile of the measured free-air impedance curve in *Fig. 2*.

Unless otherwise noted in a discussion, we will be addressing the driver mounted in a sealed box of a given volume, as it is the easiest design to consider. Horns and rear-loaded acoustic systems are more complicated to simulate and measure, and the test results are more difficult to duplicate and interpret. One advantage of a sealed box for full-range drivers is that the excursion is more limited below resonance, which can reduce distortion, particularly at higher levels.

Driven by a voltage source, we usually see anywhere from –6 to –15dB response loss at the low-frequency resonance, and a top end which increases or declines (or both) depending mostly on the cone construction and voice-coil inductance. Driven by a current source, we note that the bottom end is bumped up at resonance and the top end is increased over the performance of the voltage source.

Assume that we want to make the best frequency-response curve for a given system. This is not necessarily the best-sounding system possible, but is subject to less argument. We can trim the damping "Q" of the low-frequency rolloff knee by trying different values for R0 (L0 = 0 in this case) until it flattens out to our taste. Most of the drivers we worked with were happy with values from about 22W to about 47W. It is surprising that so little damping gave such dramatic results, but as previously pointed out, this category of loudspeaker responds well to small amounts of damping.

Usually we also want to adjust the upper midrange and high end. Often these drivers will show an upper-mid peak followed by a decline before the top end is reached. We can compensate for this by our selection of R1, C1, and L1, which can be used either to produce a shelf or a dip to flatten out the response.

With these models in mind, we measured all the sample speakers of interest, developing networks to optimally flatten out their response with a current source, and comparing this with the same speaker driven by a voltage source. We used a Vidsonics "virtual crossover" substitution box to play with the parallel loading and took both near field (an inch or so) and far field (1 meter) response measurements using Mlssa.

Again, the results here are not necessarily the best-sounding that you can get, and I am not attempting to address that as such, given the wide range of source material, listening environments, and subjective taste. Typically these exact network figures are not subjectively preferred. Usually this is because listeners tend to prefer a higher value for R1 than in *Table 1*.

If you decide to use one of these networks, consider the possibility that R1 is the minimum value and use a switch or high power (5W) potentiometer to variably increase the value to as much as twice R1. Less often, you might consider a different value for L1 in trimming the top end.

I repeat: Consider this as an experimental guide and a good starting point. Be prepared to change the values to suit your needs, and remember that these figures were obtained in stuffed sealed enclosures in a large room out from the wall. Your results will certainly vary. Also remember my comments reflect offhand reactions. Please do not treat them as reviews.

*Table 1* shows the values that resulted in improved frequency-response curves with current sources for different speakers in different-size boxes.

Now that you have an overview of the interface network, let's look at a few individual drivers in more detail.



### LOWTHER DX55

The DX55 probably has the best high-end detail among the drivers tested and also enjoyed some of the biggest improvements driven by a current source (*Figs. 3* and *4*). I consider it the best single excuse to use a current source, and it needs little or no additional damping on the low end, having a small peak in a sealed box. It ended up being the driver of choice for the KleinHorns, since the horns provide enough low-end extension to make up for the small size of the cones, keeping the excursion low.

It is not the least expensive Lowther driver at $1095/pair, but I am under the impression that the C55 will get most of this for you at $595/pair, making it more accessible. Keep in mind that subjectively you will probably want R1 at a higher value than on the table, probably between 4.7 and 10Ω.

### LOWTHER DX55 IN THE KLEINHORNS

Just as corn chips are simply an excuse to eat salsa, Lowthers provide a reason to build big back-loaded horns. These specimens are about 20′ long and 10′ high, with a 30Hz taper and a 30ft$^2$ mouth (*Photo 1*). The measurements (*Fig. 5*) are taken at 6′ because the horn doesn't develop pressure in the center until you get a ¼ wavelength away, so between the 6′ distance and the mouth size, you expect a rolloff below 50Hz, and you can see it here. Farther away, the bottom extends to 40Hz. The network for the KleinHorn is different from the others in that L0 has a non-zero value, giving low-frequency shelf attenuation—otherwise, the bass below 100Hz overpowers the mid and top.

### FOSTEX FE108E SIGMA

We borrowed this from Ed at the Horn Shop, and as far as I can tell he has the world's supply by virtue of buying up all he can get. This is one of the smaller of the latest editions of the Fostex banana fiber cones. An obvious comparison would be to the similarly sized Lowther DX55, which has much higher efficiency and a more detailed top, but the Fostex has slightly better bass extension.

It costs $85 each from Madisound. At that price, it's easy to check it out for yourself. I gather that it really prefers a rear-loaded horn—not a sealed box—but you can use it either way with good effect.

### FOSTEX FE166E IN ABBEYS

These enclosures elevate the FE166Es to a highly recommended status. Note that with a current source, you need different values, and a higher damping figure is appropriate. This seems to be a general characteristic of rear loading, and I think you can expect it with most of these drivers. You will probably also find that with a current source and R0 at 10W, a half pound of Dacron sitting in the bottom or end of the pipe smoothes out the bottom a bit. This is not necessarily needed with a voltage source, unless you think the bottom is a bit too much, which is not likely.

TABLE 1

| DRIVER | BOX FT$^2$ | R0 | L0 mH | C1 μF | R1 Ω | L1 mH |
|---|---|---|---|---|---|---|
| LOWTHER DX55 | 0.7 | | 0 | 10 | 2.7 | 0 |
| LOWTHER DX55 | Mini-Me* | 47 | 0 | 15 | 10 | 0 |
| LOWTHER DX55 | HORN* | 15 | 4 | 10 | 10 | 0 |
| LOWTHER PM2A | 1.5 | | 0 | 6.8 | 6.8 | 0.2 |
| LOWTHER DX4 | HORN* | 8.2 | 4 | 22 | 6.8 | 0 |
| FOSTEX FE108E | 0.4 | 16 | 0 | 10 | 4.7 | 0 |
| FOSTEX FE166E | 0.7 | 47 | 0 | 6.8 | 22 | 0 |
| FOSTEX FE166E | ABBEY | 10 | 0 | 10 | 10 | 0 |
| FOSTEX FE204E | 2.5 | 47 | 0 | 6.8 | 6.8 | 0.1 |
| FOSTEX FE206E | 1.5 | 47 | 0 | 15 | 4.7 | 0 |
| FOSTEX FE208E | 1.5 | 47 | 0 | 22 | 4.7 | 0 |
| FOSTEX FE208ES | 1.5 | 47 | 0 | 4.7 | 3.9 | 0.5 |
| FOSTEX FF165K | 0.7 | 47 | 0 | 6.8 | 22 | 0 |
| FOSTEX FF225K | 1.5 | 47 | 0 | 6.8 | 4.7 | 0.35 |
| JORDAN J92S | 0.4 | 47 | 0 | 6.8 | 22 | 0 |
| PIONEER BU20F20–51F | 1.5 | 23 | 0 | 33 | 10 | 0.3 |
| GOLDWOOD GW8003/8 | 1.5 | 10 | 0 | 15 | 3.9 | 0.2 |

* Mini-Me is the Lowther Mini-Medallion rear-loaded horn enclosure. Horn is the KleinHorn, a 20′ long 30Hz rear-loaded horn with a 30ft$^2$ mouth.

### JORDAN J92S

The J92S (*Photo 2*) is not considered a high-efficiency driver, at what appears to be less than 90dB/W, but it is very, very smooth and a wonderful speaker. It excels at one very difficult thing—still sounding good after the honeymoon is over. We got ours at $70 each, and I understand they are twice the price now, but they're worth it.

Magnetically shielded, they also make great computer speakers.

You'll notice here that the response curves are extremely similar (*Figs. 6* and *7*). It appears that the high-efficiency drivers make the most out of a current source, and here we don't see any particular advantage. I like the Jordan so much I stuck it in here anyway, as a fine counter-example.

### PIONEER BU20F20-51F

This speaker gets the prize for "listenable at lowest cost." It is not as efficient as a Lowther or Fostex, but it has a smooth enough character that's fairly easy on the ears. It's not as detailed, but that's alright if you like lounge music, and it can take more abuse than its more refined competition if you like to dance. Driven by a voltage source, there is a temptation to stick a tweeter on the top, and indeed, an inexpensive ¾″ dome with a 1µF cap just about does the trick.

With a current source, you obviously get more bottom end, and it requires a network to take the top end down to what you see here, which a simple RLC in parallel will accomplish nicely. The R0 value is 23Ω—low enough that if you want to play with these effects without owning a current source amplifier, you can put 25Ω of power resistor in series with the output of a 100W amp and get a Thevenin equivalent of a current source loaded by the same resistance. Will it sound as good? You have my e-mail address—let me know.

### CONCLUSION

So here we have it—a re-examination of some things that have been known for a long time but with currently available drivers. It's easy to dismiss the older ways—somewhere in the 1960s the speaker/amplifier interface took a left turn and headed off to high power voltage source amplifiers and speakers designed to lean on them for performance. A small group of iconoclastic cranks has stayed interested in these, and in the end we must recognize full-range high-efficiency speakers offer elegance, charm, and a different sort of quality.

Current sources and amplifiers with low damping offer interesting possibilities for improvement with these drivers, but they require considerable work getting the enclosures, electrical networks, and acoustic environment just right. Remember, you don't need to own a current source amplifier to put this information to work—putting R0 in series with the output of a powerful voltage source amplifier instead of in parallel with a current source will give similar results. If R0 is 47Ω or higher, you will want a big amp—300W—but in any case, be certain to use a high power resistor, say 50W or more. That said, it's all just entertainment, and I hope it inspires some of you to have some fun.

Thanks to: Kent English, Desmond Harrington, Jon Ver Halen, Terry Cain, Ed at the Horn Shop, Chris and Matt Williams.

For a complete report on the responses to individual drivers, go to www.audioXpress.com/magsdirx/aX/addenda/index.htm.

# Musicality and Distortion: A Conversation

Three audio experts discuss the role of distortion levels in what we hear.

It is with hesitation and timidity that I should cross (s)words with such an eminent audio personality as Jean Hiraga, but this hesitation and timidity is overridden by my search for technical correctness and understanding.

In his article ("Amplifier Musicality," *aX* March 2004, p. 32), Mr. Hiraga discusses the distortion spectra of various amplifiers. According to his experience, there is a correspondence between a gradual decrease of higher harmonic components and the subjective quality of an amplifier. In particular, amplifiers where the harmonic distortion spectrum does not monotonically decrease, or where specific harmonics are absent, may sound less agreeable even if the absolute distortion levels are very low.

Mr. Hiraga argues that higher distortion levels are not in themselves objectionable if the harmonic components show an orderly, monotonous decrease. Reference is made to a violin, which produces a sound that also shows many (up to 20) harmonic components. Any change in the relative amplitude of these components would alter the "color" of the violin.

One transistor amplifier, #2 (Fig. B), shows an almost total absence of harmonic components beyond the third. At the end of the article Hiraga discusses amplifier #11 (Fig. L), a single-stage WE300B, which shows a near-perfect row of gradually decreasing harmonics. Mr. Hiraga then continues, and this is where I part with him: Amplifier #11 would reproduce all the harmonics of a violin very well, while amplifier #2 would truncate the violin spectrum, absorbing it in the feedback. I am surprised by this statement, which is clearly and fundamentally wrong.

Assume for the sake of discussion that a hypothetical violin would produce a 1kHz sound plus two harmonics, each 20dB lower in amplitude than the previous one (the 2nd harmonic being 20dB below the fundamental). It is understood that the actual violin spectrum is much more rich, but just these few harmonics in the analysis will prove the point. Also assume that amplifier #2 has only a 3rd harmonic distortion component at 80dB below the fundamental (Fig. B). This amplifier #2, in reproducing the violin sound, would add a 3rd harmonic to all source components at –80dB below the original level.

It becomes more complex with amplifier #11 (Fig. L), which would add many harmonic components to each original violin component, with each additional component some 10dB lower than the previous one, and the first harmonic (the 2nd) 20dB below the fundamental. To keep this manageable, I have looked at only the first three harmonics (2nd, 3rd, 4th) produced by amplifier #11. I have tried to capture the resulting spectrum in each case in *Table 1*.

**TABLE 1**

| HYPOTHETICAL VIOLIN | AMP #2 ADDS | AMP #11 ADDS |
|---|---|---|
| 1kHz at 0dB | 3kHz at –80dB | 2kHz at –20dB<br>3kHz at –30dB<br>4kHz at –40dB |
| 2kHz at –20dB | 6kHz at –100dB | 4kHz at –40dB<br>6kHz at –50dB<br>8kHz at –60dB |
| 3kHz at –40dB | 9kHz at –120dB | 6kHz at –60dB<br>9kHz at –70dB<br>12kHz at –80dB |

One point is immediately clear: It is amplifier #11, with the rich, monotonous decreasing distortion spectrum, that alters the spectral composition of the hypothetical violin much, much more than amplifier #2, which shows predominantly 3rd harmonic distortion. Why, then, would listeners prefer (if this is indeed the case) this strongly altered spectrum to the original one? One theory could be that although amplifier #11 alters the spectrum much more, it better preserves the spectral *shape* than amplifier #2.

*Table 2* shows that this also is a false assumption: Amplifier #2, with its seemingly ugly 3rd harmonic spike, has only a very, very benign impact on the violin spectrum. Amplifier #11, however, is terrible. Not only does it significantly alter the spectrum *content* (it adds 4, 6, 8, 9, and 12kHz components, which are totally absent in the original), it also significantly alters the spectrum *shape*. The 3rd harmonic at the output of amp #11 is more than ten times as strong as in the original of amp #2. The ugly 9th harmonic is fully 50dB stronger at amp #11 than at amp #2.

It seems to me that the conclusion is inescapable. Amplifier #11, for all of its nicely, monotonously decreasing harmonic spectrum, totally messes up the violin. If listeners nevertheless prefer a type #11 amplifier, it is not because it more faithfully preserves the original spectrum, because it doesn't. It must be because listeners prefer a heavily mutilated spectrum to the original source.

Jan Didden
Luxembourg

Regarding "Amplifier Musicality," by Jean Hiraga, I'll start with the article's fundamental premise: "Even though the distortion in an amplifier may be at extremely low levels, the content . . . must conform to the laws of harmonic levels in order to give subjectively musical reproduction." (These "laws" are refer-


Transformers & Enclosures
In Stock...
Torodial Power:
13 sizes (15 - 1500 VA)
6 VAC - 240 VAC
"Classic" Tube Output:
Single Ended (up to 75 watts),
Push-Pull and Potted (up to 280 watts)
Enclosures:
Aluminum & Steel Chassis,
Diecast Aluminum, Plastic
and Rack Mount
Filament & Low Voltage
Power Transformers:
Open & Enclosed
"Classic"
High Voltage Transformers:
Plate, Plate/Filament Combo
& Filter Chokes (Open & Enclosed)
Contact us for free catalogs & a list of stocking distributors
HAMMOND
MANUFACTURING TM
256 Sonwil Dr. - Cheektowaga, NY 14225 USA
Phone: (716) 651-0086 Fax: (716) 651-0726
394 Edinburgh Rd., N. - Guelph, Ontario N1H 1E5 Canada
Phone: (519) 822-2960 Fax: (519) 822-0715
www.hammondmfg.com

enced to Olson's observation that pure tones don't sound musical, but a regularly reduced harmonic structure can.)

That an amp's distortion, if well below audible threshold, must conform to "musicality" should be obviously questionable. Consider "Single-Ended vs. Push-Pull" by Scott Frankland (*Stereophile*, Jan. 1997): He notes a resemblance between the distortion spectra of a certain amplifier and the ear, and then comments that this ". . . does not in any way justify the presence of distortion in playback systems."

Mr. Hiraga shows a sample violin note spectrum, then says, "The slightest modification of the relative levels will change . . . the tonal color . . . making a Stradivarius sound like a very cheap violin . . ." Really? Moving one's head an inch (let alone bowing the violin differently, and so on) will more than slightly change the spectrum! But a Strad still sounds like a Strad, because something in the harmonic patterns—how they dynamically change, and in harmonic sounds—is identifiable.

Finally, Mr. Hiraga says about amplifier 11: "A violin, in particular, is reproduced very well as harmonics 2 to 20 pass without interference in their relative levels . . ." Unfortunately, with about 9% THD (Fig. L, p. 35), a pure organ stop will also be reproduced with violin-like harmonics!

Some of the amplifiers have all harmonics below 70dB. At moderate SPL the ear can generate 1% THD (–40dB). So it seems ludicrous to attribute anything to the structure of such an amp's distortion. Of course, static single-tone distortion bears about as much relation to that produced by an amp delivering music to a reactive load as kicking tires does to driving the car!

If the spectra shown maintain their shape but the levels increase greatly in the real world of complex music waveforms and reactive speakers (a big "if"), then the added harmonics may blend in with, or fill in recorded losses of, natural instrument harmonics. But even if this is possible, to generalize a correlation of the figures shown to overall "musical performance" is absurd.

Dennis Colin
Gilmanton I.W., N.H.

*Jean Hiraga responds:*

*High fidelity is the closest possible reproduction of the original. With regard to an amplifier, this means that the output waveform must in all respects be the amplified exact replica of the input signal. Subjective high fidelity is a different matter because it must take into account personal appreciation of a listener. This aim is attained when the listener declares to have the same aural sensation that he would have had in a concert hall or in a studio.*

*The ear remains the judge of fidelity, faced with the various types of distortion such as frequency distortion, phase distortion, transient distortion, IM and IIM distortion, scale distortion (intensity distortion), frequency modulation distortion, amplitude distortion, and harmonic distortion.*

*If measured harmonic distortion can take into account individual levels, aural sensation of harmonic distortion must refer to fundamental aspects of tone quality that are sonance (the successive presence of fusion of timbre continuously changing, the pitch, and the sound intensity in a tone as a whole) and the timbre (defined by the simultaneous presence of fusion of the fundamental frequency and its partials at a given moment). Since 1924, R.L. Wegel and C.E. Lane have found effects of auditory masking of pure sound by harmonics or by other tones[1].*

*It is well known that harmonics added to a pure tone are not necessarily displeasing, since voices, sounds of various origins, and music instruments produce notes made of a complex family of harmonics. Wegel and Lane were also among the first to discover that harmonics of defined intrinsic levels added to a pure tone can be heard as a pure tone by successive mask effects and multiple harmony incidence.*

*Concerning the same matter, it is important to take into account the fundamental work of Carl E. Seashore[2] in his book* Psychology of Music. *In my article, Fig. 1 shows a sample of violin and its relative levels. The original article, published 25 years ago in* Nouvelle Revue du Son, *mentioned that that figure was from* Psychology of Music.

*In his book titled* Loudspeakers, The How and Why of Good Reproduction, *G.A. Briggs of Wharfedale Wireless Works refers to the same work of Seashore as follows "Harmonic*

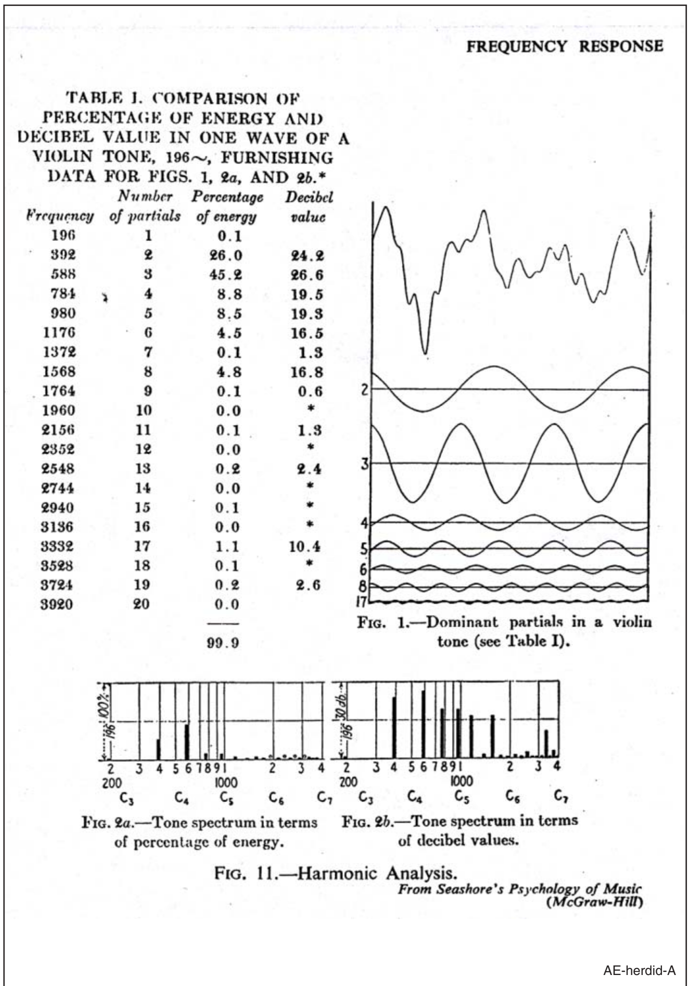

FREQUENCY RESPONSE

TABLE I. COMPARISON OF PERCENTAGE OF ENERGY AND DECIBEL VALUE IN ONE WAVE OF A VIOLIN TONE, 196~, FURNISHING DATA FOR FIGS. 1, 2a, AND 2b.*

| Frequency | Number of partials | Percentage of energy | Decibel value |
|---|---|---|---|
| 196 | 1 | 0.1 | |
| 392 | 2 | 26.0 | 24.2 |
| 588 | 3 | 45.2 | 26.6 |
| 784 | 4 | 8.8 | 19.5 |
| 980 | 5 | 8.5 | 19.3 |
| 1176 | 6 | 4.5 | 16.5 |
| 1372 | 7 | 0.1 | 1.3 |
| 1568 | 8 | 4.8 | 16.8 |
| 1764 | 9 | 0.1 | 0.6 |
| 1960 | 10 | 0.0 | * |
| 2156 | 11 | 0.1 | 1.3 |
| 2352 | 12 | 0.0 | * |
| 2548 | 13 | 0.2 | 2.4 |
| 2744 | 14 | 0.0 | * |
| 2940 | 15 | 0.1 | * |
| 3136 | 16 | 0.0 | * |
| 3332 | 17 | 1.1 | 10.4 |
| 3528 | 18 | 0.1 | * |
| 3724 | 19 | 0.2 | 2.6 |
| 3920 | 20 | 0.0 | |
| | | 99.9 | |

FIG. 1.—Dominant partials in a violin tone (see Table I).

FIG. 2a.—Tone spectrum in terms of percentage of energy.

FIG. 2b.—Tone spectrum in terms of decibel values.

FIG. 11.—Harmonic Analysis.
*From Seashore's Psychology of Music (McGraw-Hill)*

AE-herdid-A

**FIGURE A: Harmonic analysis (from Seashore's *Psychology of Music*).**

**TABLE 2**

| FREQ | VIOLIN | OUTPUT AMP #2 | OUTPUT AMP #11 |
|---|---|---|---|
| 1kHz | 0dB | 0dB | 0dB |
| 2kHz | –20dB | –20dB | –14dB |
| 3kHz | –40dB | –39.9dB | –18dB |
| 4kHz | | | –34dB |
| 6kHz | | –100dB | –48dB |
| 8kHz | | | –60dB |
| 9kHz | | –120dB | –70dB |
| 12kHz | | | –80dB |

analysis: At this point, it may be of interest to see what actually constitutes a musical tone heard by the ear. This is found out by harmonic analysis. An example from Seashore's Psychology of Music gives the tone spectrum of a open G string of a violin played with medium intensity. It will be noticed that there is only 0.1% of energy of the fundamental frequency of G 196, and this shows why the fundamental can be omitted from reproduction without destroying the pitch of the note. This fundamental is established by the difference-tone of 196 which occurs between the overtones. The top wave of the figure is an oscillogram of the full tone, which was analyzed up to 20 partials, covering a total frequency range of about four octaves. When one reflects that a single note has such a complicated structure, it is difficult to imagine the thousands of wave formations produced by a full orchestra . . ." (Fig. A).

Seashore takes into account the same matter in another of his works, "In Search of Beauty in Music." In "The perception of Music, psychological origins and development of the sense of Tonality," authors Robert Frances and W. Jay Dowing developed a similar point of view.

It is important to keep in mind that Wegel, Lane, and Seashore didn't have at their disposal enough pure sound generators to develop their investigations more completely. In the '40s, a lot of works by authors such as H.A. Hartley[3], IEE Radio Section[4], and, more particularly, H.F. Olson[5] in "Elements of Acoustical Engineering," related the importance of harmonic distortion spectra classed in categories such as "perceptible, tolerable, and objectionable distortion." In his single-stage 2A3 triode amplifier, Olson stated that he obtained distortion levels of –30dB, –50dB, and –70dB for 2nd, 3rd, and 4th harmonics. The last –70dB, or 0.03% corresponding to the audio generator self-distortion, explains why Olson was unable to measure respective levels of 5th and higher order harmonics.

Also, keep in mind that in amplifiers, classical push-pull operation leaves the third and odd harmonics predominant. In tube circuits, if the grid modulation level is increased to give the same value of total distortion as on the single triode circuit, the high order harmonics become more significant and unmasked by first harmonics. If this remark applies to sounds of relatively slow attack and decay, this effect is largely overshadowed by the more prominent influence of transient sounds, including a large part of inharmonic content added to the harmonic content, as explained in an article of A.W. Ladner[6] titled "The analysis and synthesis of musical sounds."

A unique faculty of the ear is to establish timbre personality and fundamental notes by

**TABLE 3**

| HARMONIC ORDER | FREQUENCY | MUSICAL SCALE (NATURAL) | EFFECT |
|---|---|---|---|
| 2nd | 500 | $C^{i}$ | |
| 3rd | 750Hz | G | |
| 4th | 1kHz | $C^{ii}$ | |
| 5th | 1.25kHz | E | |
| 6th | 1.5kHz | G | |
| 7th | 1.75kHz | – | Dissonant |
| 8th | 2kHz | $C^{iii}$ | |
| 9th | 2.25kHz | D | Dissonant |
| 10th | 2.5kHz | E | |
| 11th | 2.75kHz | – | Dissonant |
| 12th | 3kHz | G | |
| 13th | 3.25kHz | – | Dissonant |
| 14th | 3.5kHz | – | Dissonant |
| 15th | 3.75kHz | – | Dissonant |
| 16th | 4kHz | $C^{iv}$ | |
| 17th | 4.25kHz | – | Dissonant |
| 18th | 4.5kHz | D | Dissonant |
| 19th | 4.75kHz | – | Dissonant |
| 20th | 5kHz | E | |
| 21st | 5.25kHz | – | Dissonant |
| 22nd | 5.5kHz | – | Dissonant |
| 23rd | 5.75kHz | – | Dissonant |
| 24th | 6kHz | G | |
| 25th | 6.25kHz | $G_{\#}$ | Dissonant |

successive analysis of harmonics starting with the highest order ones, as stated in various works of S.S. Stevens, Von Bekesy, and E. Zwicker. This is why our brain has the ability to reconstitute a fundamental note from listening to its partials (this was tested by Carl E. Seashore). If the amplifier induces partial or total suppression of some harmonics among those forming THD, this is taken in account by the ear which will detect a corresponding timbre.

This does not exclude the masking effects, more particularly, those for which harmonics relations and relative levels conduct to hear a unique pure tone, referring to extensive works conducted by H. Sakai[7] in his "Harmony and distortion" paper published in 1980 or by a Japanese group titled The Fukuoka Distortion Society.

It is, however, important to say this "works" only for THD values under about 3%, confirming the experiments conducted by H.F. Olson with 2A3 single-stage triodes.

The study of acoustics, psychology of music, and violin sounds may be interesting ways to understand the subjective effect of some harmonics in relation to naturalness and beauty of sound. But for easier understanding, refer to Table 3, which shows a fundamental frequency of 250Hz and its harmonics from 2nd to 25th, as well as related dissonant effects.

In the field of violin study, works of world authorities such as Carleen Maley Hutchins[7] and Fernand Dacos (during his work at Conservatoire of Liege, Belgium, in the 50s and 60s, he had the opportunity to analyze more than 1,500 violins) detailed with accurate measurements harmonic relations and respective levels, including the effect of the shape of audibility curves to the ear and its maximum sensitivity of around 3kHz. Their conclusions are similar to those found by A.W. Ladner. A third octave analyzer can show that most sound energy produced by musical instruments is concentrated between 50Hz and about 1.5kHz.

Harmonics are spread out to the upper limits of audibility, with respective levels decreasing very fast above 5kHz, in areas where "bad musicality" amplifiers produce irregular high-order harmonic distortion unmasked by first-order ones.

Concerning harmonic distortion spectra shown in my article, I must add the following:

- In amplifiers, regular decay of harmonic distortion spectra is not always proof that the circuit doesn't selectively absorb some harmonics, especially in the case of unconventional complicated circuits.
- Slight asymmetric switching distortion induced by a push-pull circuit or by poor-quality output transformers may generate a series of harmonics that can be the cause of confusion between good- and bad-sounding amplifiers.
- A harmonic distortion spectrum with a regular decrease may change with fundamental frequency and level and modify sound quality accordingly.
- In a classic amplifier made of several stages, each stage has its proper characteristics—input/output transfer curve, linearity, harmonic distortion—and is affected by other kinds of distortion in static (constant level, fixed frequency) or in dynamic/transient mode. In dynamic and transient mode corresponding to normal music and sound conditions, these phenomena and nonlinearities are not simply added or subtracted from others. They are the cause of very complex transient cumulative effects of nonlinearities that change with each tiny portion of amplified and reproduced signal on real (complex) loads.

*Dennis Colin raises an important point concerning an amplifier's delivery of sound to a reactive or to a real load and effective back energy (CEMF energy) released by the loudspeaker. An article of mine published in* Revue du Son & du Home Cinema[8], *concerns an experiment that was conducted 20 years ago by a group from Tokyo Musashino University in Japan. The experiment simulated back energy of a loudspeaker comparing its effects on a pure or a real complex load, while the amplifier under test was stimulated by another frequency.*

*The published results—limited (because of lack of space) to four amplifiers—revealed very interesting dynamic comportments of each amplifier in that condition. In that article, Fig. 1 shows original signal spectra with pre-filtered 50Hz applied directly to the load and the 1kHz signal applied to the input of the amplifier; Fig. 2 illustrates a good example of an amplifier not influenced by this condition; Figs. 3* and *4 show two other examples of amplifiers exceptionally good under normal measurement conditions, each using different complex active feedback, showing on that test important generation of harmonics that would be a good explanation concerning their "inexplicably harsh and brittle-sounding" output; and Fig. 5 shows a single-stage amplifier (801A), with formation of harmonics and subharmonics with "paradoxically" a quite tolerable subjective distortion effect due certainly to regular decay envelopes.*

*A lot of matters that cannot be easily explained still remain, especially because we are still unable to access ultra-fast measurements in real working conditions, underneath the musical signal. It is now an established fact that significant subjective differences of sound quality can be noted between several kinds of resistors, capacitors, and coils (of loudspeaker networks), even if their harmonic distortion spectra are extremely low (–100dB or below), or 20 to 30dB below floor at audition.*

*Concerning this point, note that the form of harmonic distortion induced by such components is frequently similar—or very close to—the same form of harmonic distortion "lifted" in audible levels. A good example is the case of some high-priced ultra-high stability resistors using the method of tiny matched positive and negative thermal elements that sound worse than a version using, for example, a pure tantalum oxide single resistive element.*

*I prefer not to enter into the domain of "sound of cables" or into the matter of "sound of passive components," because I know the influence of my first articles published on this matter in* Nouvelle Revue du Son *and in* L'audiophile *review from 1976 and 1977. Conventional measurements, made with even very sophisticated and ultra-low distortion meters, are still producing mostly insignificant results, because they are located largely under the floor of audibility.* ❖

## REFERENCES


1. R.L. Wegel and C.E. Lane, "The Auditory Masking of Pure Tones," *Physical Review*, 23, 266-285, 1924.

2. Carl E. Seashore, "Psychology of Music," McGraw-Hill, New York.

3. H.A. Hartley, "Aesthetics of Sound Reproduction—High Fidelity or Judicious Distortion?" *WW*, July/August 1944.

4. "IEE Radio Section, Distortion—Does it matter?," *WW*, January 1949.

5. H.F. Olson, "Elements of Acoustical Engineering," *Van Nostrand*, 1947.

6. A.W. Ladner, "The Analysis and Synthesis of Musical Sounds," *Electronic Engineering*, October 1949.

7. H. Sakai, "Harmony and Distortion", N.H.K *Journal*, 1980.

8. J. Hiraga, "La Distortion dans tous ses états," *Revue du Son* & *du Home Cinema*, n°279, November 2003.

# The Adventure Begins

Excerpted and condensed from Chapter 1 of Ray Alden's forthcoming book, *Speaker Building 201.*

## STEPS TO DESIGN AND BUILD A PAIR OF SPEAKERS

First you need to decide which drivers you intend to use. Drivers are the parts that actually move the air; that is, larger-size woofers for the bass, mid-size drivers for the midrange, and tweeters for the high frequencies. These three parts are the gateway to building a *three-way* speaker system. To build a *two-way* system you need only a tweeter and a mid-bass unit, which range from 4″ to 8″.

Except for one horn-loaded system, this book will restrict discussion to closed box and vented systems, which are the most accessible and straightforward for beginners. Eventually, readers may wish to continue beyond the basics to explore passive radiators, electrostatic speakers, transmission lines, or more complex horn-loaded systems.

Each driver has its own special characteristics, most quantified by numbers described in its *parameters*. Other characteristics, such as *frequency response* and *impedance*, are typically shown graphically. For most drivers these days, you can find these important numbers and graphs on a manufacturer's specification (*spec*) sheet. However, many builders prefer to measure them directly, which is often more accurate. Placing these numbers into equations or entering them into special computer programs will provide the reliable answers needed in the design process.

The two main ingredients needed in speaker design are finding the correct volume of the cabinet enclosure and finding which electrical parts are needed to build the *crossover*. A crossover is an electric device that uses resistors, capacitors, and inductors to filter out certain frequencies going to the drivers. For example, a properly designed crossover makes certain that low frequencies do not reach the tweeter in sufficient amounts to either cause harm or unpleasant sound effects. At the same time, the crossover allows high frequencies to go to the tweeter. The crossover will not only protect each driver but, if well chosen, will allow exactly the correct band of frequencies to arrive in the proper proportion to each driver of the speaker system.

*Figure 1* shows a transparent speaker system with box, drivers, and crossover. A simple crossover is shown to the left in this figure between the amplifier and drivers. Once you know the cabinet size and the crossover values, you are then ready for construction. You'll need to do some hand work, even if only to assemble the parts of a kit. However, most projects have fairly involved woodworking and wiring, including the use of miter joints, routing recessed holes for the drivers, and the soldering of a complicated crossover.

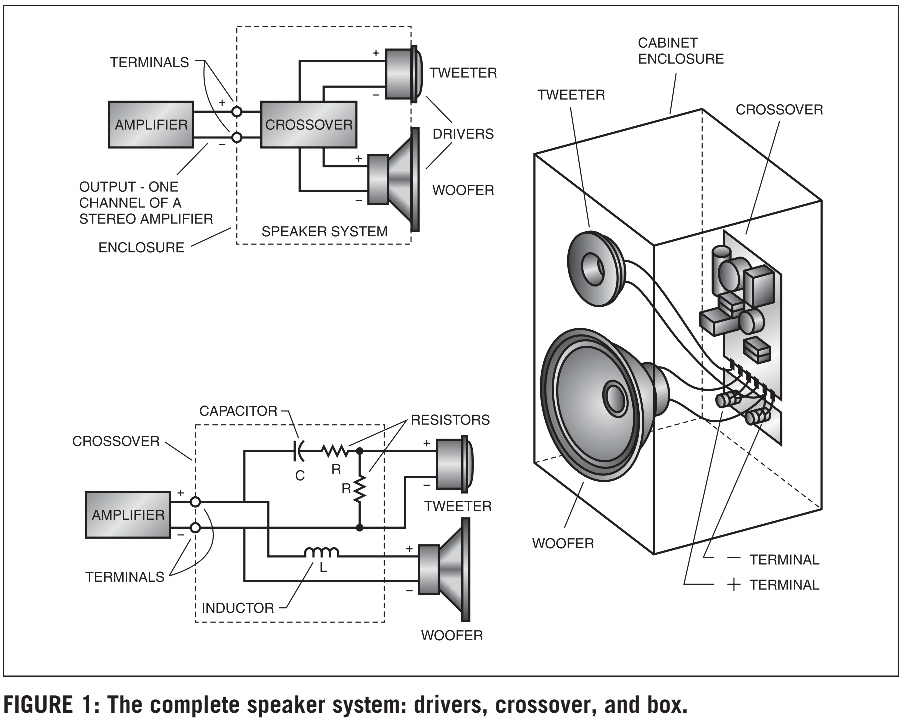



**FIGURE 1: The complete speaker system: drivers, crossover, and box.**

## DEFINING THE DRIVERS YOU NEED

Once you have decided on your budget, you will be looking for drivers that can work together to give you the smoothest integrated frequency response. What follows is the information you need to make an intelligent choice.

To begin, look at the manufacturer's *spec* sheet. Here you must choose the features that produce the "best of all possible worlds" for your personal preferences, in which case some characteristics become more important than others. You might be an audiophile who wants the most exacting fidelity or you could be a DJ who needs to produce a very robust "bulletproof" system. The DJ may be willing to sacrifice ultra-smooth frequency response or precise imaging in order to have a speaker that is very efficient, one capable of long cone excursions and able to take large amounts of power. These types of needs form the basis of what is known as the "Pro Sound Market."

Although not necessarily mutually exclusive, it often is the case that a driver is optimized specifically for one objective or the other. While most readers will find this book more geared toward the audiophile, nonetheless at several junctions pro-sound applications will be mentioned. Whatever your situation, an under-

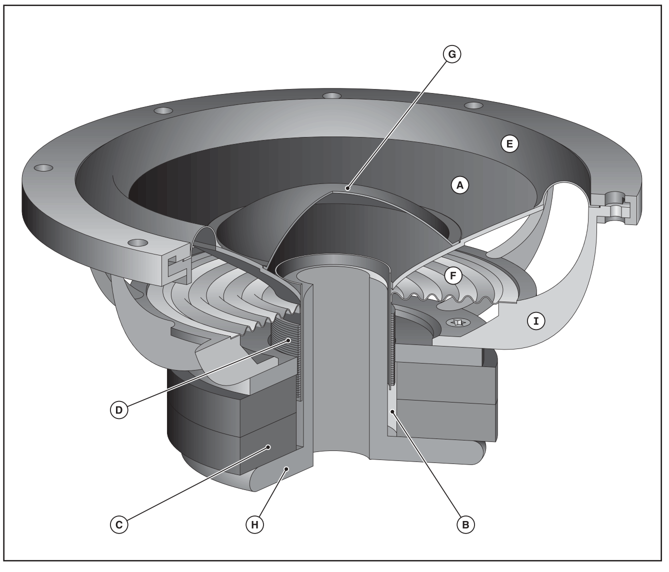



**FIGURE 2: A cut-away section of a moving-coil driver showing the various parts.**

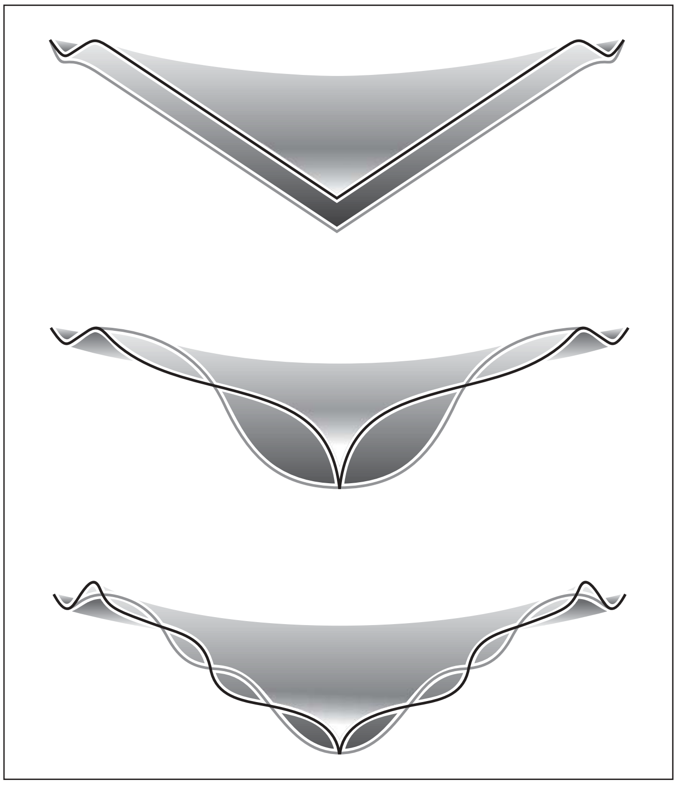


**FIGURE 3: "Break-up" modes in a driver cone.**

standing of the parts that make up a driver and the numbers that determine driver behavior are paramount. Paving the way for this understanding are some basic notions embedded in the history of the development of electricity and physics. Technology keeps improving drivers so that the most recent are often the best.

## NO BARGAIN

Should you jump to purchase a driver advertised in the following way:

"For sale: 12″ woofer, goes from 34Hz to 5kHz at 8Ω, $85, excellent value."

This blurb gives less information than it seems. The only thing you could be sure about is the diameter of the driver. Saying this woofer operates from 34Hz—5,000Hz (Hz is short for *hertz*, the frequency in *cycles per second*) gives you no idea of the SPL (*sound pressure level* measured in dB or *decibels*) at which the driver functions when a certain number of *watts* of power is input. It may well produce a sound at 34Hz, but perhaps at a level so low you cannot hear it.

Likewise, although the range is claimed to go up to the relatively high frequency of 5kHz (the "k" represents *kilo*, or 1000), can you actually hear it? What about the band of frequencies in between? The response might be very ragged, with numerous drops and high peak resonances. Likewise, specifying 8Ω is only a nominal value, and as has been mentioned, drivers (with the possible exception of ribbon designs) do not behave like resistors at all.

The word i*mpedance* is used rather than resistance to describe the complex nature of resistance in a driver (resistance and impedance are both measured in *ohms*). The impedance of a driver changes as the signal frequency changes, which is why the graph of a driver's impedance will be a curve rather than a straight line. Even the mention of a 12″ woofer does not mean it is capable of driving significant amounts of air without distortion. Quite possibly the cone may have a very restricted motion or a minuscule magnet on the back more suited for use in a tweeter. A spec-sheet, with a description of features, frequency response, and impedance graphs, would be of immensely more help in making a decision than the info in the for sale ad.

## A CUT-AWAY DIAGRAM SHOWING THE PARTS OF A SPEAKER DRIVER

In *Fig. 2* you see the essential parts that make up a speaker driver. Many of these parts have a direct bearing on the quality of sound the driver contributes in a speaker system. Let us examine some of the parts referred to in the figure.

- A—the Cone. The part that actually moves the air. Technology has continually improved cone materials, always striving to attain a better ratio of extremely light material to very rigid fabric. The attempt is to find a textile composition that will permit the driver motor to move the cone quickly and efficiently yet control cone "break-up" modes that can cause nasty resonance peaks. *Figure 3* shows how a cone can flex as frequency increases . . . At first, paper

| Complete Specs Including Extrapolated Data From Bassbox | | |
|---|---|---|
| | SI units | U.S. units |
| Mechanical Parameters | | |
| Fs | 28.31 Hz | 28.31 Hz |
| Qms | 9.547 | 9.547 |
| Vas | 38.09 L | 1.345 cu.ft. |
| Cms | .237 mm/N | .0415 in/lb |
| Mms | 133.9 g | 4.723 oz |
| Rms(calc) | 2.485 kg/s | 5.478 lb/s |
| Air Gap Height | 8 mm | .315 in. |
| V.C. Diameter | 50.8 mm | 2.0 in. |
| V.C. Length | 38.1 mm | 1.5 in. |
| Xmax | 16 mm | .63 in. |
| Xmech(est) | 24 mm | .95 in. |
| Diameter | 208 mm | 8.18 in. |
| Sd | .034 sq. m | 52.7 sq.in. |
| Vd(calc) | 538 cc | 32.83 cu.in. |
| Magnet Weight | 2.38 kg | 84 oz. (5.25 lbs.) |
| Electrical Parameters | | |
| Qes | 0.427 | 0.427 |
| Re | 2.467 ohms | 2.467 ohms |
| Le | 2.48 mH | 2.48 mH |
| Z(nom) | 4 ohms | 4 ohms |
| BL | 11.74 | 11.74 |
| Power (RMS) | 300 watts | 300 watts |
| Power (Max) | 425 watts | 425 watts |
| Electromechanical Parameters | | |
| Qts | 0.407 | 0.407 |
| no(efficiency) | 0.195% | 0.195% |
| 1-W SPL(calc) | 85.1 dB | 85.1 dB |
| 2.83V SPL(calc) | 90.21 dB | 90.21 dB |

**FIGURE 4: The spec sheet for a Titanic 10MKII woofer.**

was the universal material used, but now bextrene, polypropylene, pearl mica-injected polymers, woven fiberglass, woven carbon fiber, metal alloys, aerogel™, Kevlar™ thin ceramic material, and even banana fiber are used in woofers and midranges.

Some manufacturers are returning to paper, but using special types either treated or reinforced with synthetics such as Kevlar. Dome tweeters will sometimes use aluminum, titanium, silk, or other treated fabric to reproduce the high-frequency domain. As you might suspect, each of these materials has its own advantages and disadvantages. For example, metal alloys often sound exceptional over a certain range but can have an exaggerated response at high frequencies, something a designer should take into account by appropriate filtering in the crossover design.

Size often matters. For example, cone diameters have an effect on high-frequency reproduction. Although large cone areas can move more air at lower frequencies, these same large areas meet with increasing air resistance when attempting to replicate higher frequencies. Small area diaphragms such as those found in tweeters are far better at both reproducing the high-frequency range and providing broad dispersion. For similar reasons, medium-size cones are used for the midrange of sound reproduction.

Much of the search for better cone materials attempts to achieve excellent *transient response* from a speaker driver. Good transient response is the capacity of a driver to accurately reproduce the sharp staccato attack of musical notes that come and quickly go without continuing to "ring" like a bell after the note is gone. If a driver cannot control itself from continuing to sound the long-departed spent note, noticeable muddiness occurs.

- B—the Gap. This is the critical opening between the front plate and the pole piece. The pole piece is made of iron pieces which become the north and south pole for the surrounding cylindrical magnet. In this gap a powerful permanent magnetic field is concentrated, poised to interact with any electromagnetic field produced by an alternating current sent through the voice coil. Once the two fields interact to produce a force on the wire in the direction governed by the *right-hand rule* of vector geometry, the voice coil (and attached cone) will be set into a motion perpendicular to the permanent magnetic field.

- C—the Magnet. Typically made of a ferrous ceramic material—(strontium-iron) or sometimes alnico (aluminum-nickel-cobalt)—the size and weight of the magnet has a direct bearing on the strength of the motor system. Sometimes a manufacturer will make use of a double (as seen in *Fig. 2*) or triple magnet system to increase the strength of the magnetic field. Three of these driver types are used in the speaker systems of Chapter 11.

In the small space allotted for tweeter magnets, manufacturers will resort to expensive rare earth materials such as neodymium-iron-boron to maximize the magnetic field. Rare earth magnets can produce a magnetic strength as much as 35 times greater than ceramic ferrous types. The maximum energy product of a magnet, in cgs units, is a combination of its residual induction strength in gauss and its coercive force in oersteds.

- D—the Voice Coil. The voice coil is made of insulated wire, through which the signal will flow, wrapped around a cylinder called the *former*. This cylinder is critically aligned and inserted into the driver's gap. Along with the magnet, this forms the basis for the driver's motor. The strength of the voice coil comes essentially from the length of wire wrapped around the former.

The geometrical position of the voice coil in the gap determines how far the cone can safely move in and out of the gap, designated as Xmax (usually given in millimeters, mm). This amount can be as much as 16mm in long throw woofers and this, along with a sizable voice-coil diameter for dissipating heat, gives some indication of how resilient a driver will be when driven hard. Sometimes a former, using a material such as Kapton™, is coated to aid in dispersing heat build-up.

- C–D—the Driver Motor. When the voice-coil wire length (l) is multiplied by the magnetic field density (B), the Bl product gives an idea of the force that can be exerted on the cone in Tesla meters per newton. When the Bl product is multiplied by the amount of current (*i*) flowing in the voice coil, the exact force can be calculated. In general, the greater the Bl product and the lighter the cone in combination with an appropriate suspension system, the more efficient the speaker will be at producing sound. In the 1950s, when many drivers were inefficient and low-powered tube amplifiers abounded, horn-loaded speaker systems were the rage for producing hefty sound pressure levels.

As in Edison's early phonograph, a horn can transform a driver's high sound pressure output by gradually channeling it through a controlled air column so as to more efficiently match the driver's high acoustic impedance to the air's low acoustic impedance.[1] With today's improved driver motors and amplifiers, horns are not a necessity. But successful home system horn designs are made to this day. Horn-loaded systems are often seen in PA systems and found in large spaces such as auditoriums and clubs.

- E—the Surround. The part of the driver's suspension system in which the cone terminates. It is usually made of rubber or foam. Many older commercial drivers used foam material in the surround which has crumbled over the years. "Re-coning kits" are now available to correct this failure.

Fortunately some modern foam contains an anti-UV agent to reduce deterioration. Since cone vibrations terminate at the surround, if it can quickly damp these and thus reduce ringing, it provides a welcome improvement. Some surround materials such as norsorex™, if made into a ball and dropped, has very little bounce. Such materials have excellent self-damping characteristics.

- F—the Spider. Attached to but hidden behind the cone is a wavy ridged piece of material (often linen) that resembles ripples frozen in a pond. Its primary function is to provide a critical restoration force to the pushing and pulling of the cone. It greatly affects the resonant frequency of the driver, as do the cone and voice-coil mass. Those planning to play speakers at high SPLs or intending to build speaker systems for DJ or club work might look for a feature known as a "progressive spider" in the suspension system. This spider type adds an important characteristic to drivers; as a voice coil travels beyond its Xmax range, this suspension system puts a progressively stronger brake on the motion of the cone.
- G—the Dust Cap. The material of this dome is designed to prevent small particles such as dust from entering the gap. However, if the cap is made of solid material, air compressions and rarefactions will form in the gap area. But if the cap material leaks air, then making an airtight cabinet can prove difficult. By re-thinking the design of the back plate, engineers have found one way to resolve this challenge.
- H—the Back Plate. This piece is sometimes vented through a hole in the rear. This hole helps relieve the air pressure caused by a solid dust cap and releases any heat build-up in the gap.
- I—the Frame. Integrates and holds the suspension system, cone, and the driver motor together. As you might suspect, rapid and vigorous motion of the cone transfers vibrations to the frame. Many speaker drivers use a simple stamped steel frame. Strong as this is, more expensive units often use a more massive magnesium or aluminum cast frame, producing a supporting skeleton more immune to pulsating cone effects.

## SPEC SHEETS

*Figure 4* shows the specification sheet for a Titanic MKII 10″ woofer, used in Chapter 11 in one of the designed systems. If you are new to speaker building, this myriad of letters and subscripts must seem a muddle. What is important? What do they mean? How do I use them?

These parameters are important, and many must be used in order to find out what box size is needed or even what type of box to use. They are called Thiele/Small parameters, named after Neville Thiele and Richard Small, two engineers whose technical papers explained the alignments and calculations needed to find the size of closed and vented speaker cabinets. ❖

Speaker Building 201 *is now available from Old Colony Sound Lab, PO Box 876, Peterborough NH 03458, 603-924-9464, Fax 603-924-9467, e-mail custserv@audioXpress.com.*

**REFERENCE**

1. Abraham Cohen, *Hi-Fi Loudspeakers and Enclosures*, Hayden Books, 1968.

# High-Quality Tube Type Control Unit

This author's latest design is a control unit that offers "lifelike" concert-hall performance for a modest cost of $160. **By Joseph Norwood Still**

**PHOTO 1: The control unit (external view).**

This control unit design features high-perveance tubes in the phono and line amplifier. The phono section contains two 6N1Ps and one 6922, 33mu triodes. The phono amplifier has an output voltage of 0.68V RMS with an input voltage of 3mV/1kHz. The line amplifier uses three 5687s with a mu of 12; the 5687 is equivalent to a paralleled 6SN7. The 5687s are connected in a paralleled SRPP circuit providing a flat frequency response of 20Hz to 40kHz.

The 5687 SRPP circuit contains a "bass enhancement control" that uses a frequency limiting capacitor and inverse negative feedback to adjust (decrease) the level of the frequencies from 400Hz to 20kHz but has no effect on the frequencies from 10Hz to 100Hz. This control is especially designed for tone control "haters." The control only enhances bass frequencies a maximum of 6dB by changing the level of inverse negative feedback. This bass enhancement control is designed to convert tone control "haters into believers," at least with this circuit.

I never use tone controls, but my son Stephen expressed an interest and need for modifying the sound when playing records. I knew the traditional tone controls were not the way to go, as I tried them all and was very aware of their limitations. I gave this problem quite a lot of thought and arrived at the solution by thinking "outside the box." As mentioned, I didn't enhance the bass frequencies but instead used negative feedback and a frequency selective capacitor to reduce the level of the 400Hz to 20kHz frequencies.

The variable feedback control also had the dual purpose of injecting a small amount of negative feedback at the user's discretion. This control, with its very subtle means of simultaneously varying frequency and applying negative feedback, gave the result a discriminating audiophile can accept.

The control unit uses a total of six tubes three in the phono section and three in the line amplifier section. A commercial unit using the same number of tubes as the unit described here would typically sell for $3,000, which is a fair price when you consider that constructing this unit requires a minimum 100 man-hours. This all-tube control unit will be competitive with or outperform some of the most expensive phono/line amplifiers. The three 5687s used in the line amplifier in a paralleled SRPP circuit normally would use four tubes, but I used two tubes (each section connected in parallel) for the active amplifier device and separate sections of a single 5687 as the load and power control. Thus, I eliminated one 5687.

**Of interest**: The phono amplifier in this article does not use high-mu 12AX7s or 12AT7s, but rather linear, low noise, high-perveance 6N1Ps and a 6922, all with a mu of 33. During the early days of the introduction of magnetic phono cartridges in commercial radios, the 12AX7s were developed to eliminate a medium mu tube and thus save the radio manufacturers money. The implementation of the 12AX7 was based on cost constraints and not technical excellence. Many manufacturers who employ the 12AX7 for costly hi-fi use do not seem to be aware that they adapted a low-cost concept in their product. To provide technical excellence in phono amplifier design and ensure a superior sound, vacuum tube phono amplifiers should be limited to tubes with a maximum mu of 40.

## FEATURES

The control unit (*Photo 1*) has a remote 12V AC/8A power transformer ($25) and a chassis-mounted transformer ($7.99). The 12V AC power transformer is locat-


ABOUT THE AUTHOR

Joseph Norwood Still, retired from the electronics industry, is designing affordable high-quality audio amplifiers for the dedicated audiophile. He thoroughly enjoys this hobby and is especially rewarded with many pleasant interchanges with dedicated, resourceful audiophiles. He lives in BelAir, Md.

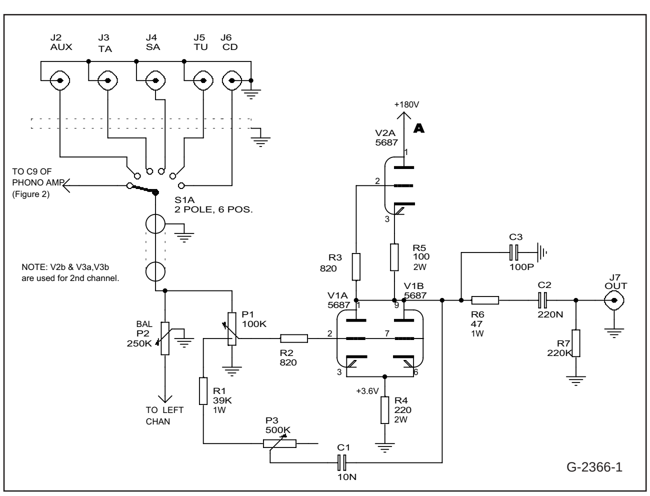




**FIGURE 1: Line amplifier schematic.**

ed remotely from the control unit chassis to ensure low-noise operation. This 12V AC transformer provides power to operate a 12V AC (input) to 120V AC (output) chassis-mounted power transformer and a full-wave, bridge-rectifier to furnish power to the filaments of the control unit.

The 120V AC output of the chassis-mounted transformer "feeds" a full-wave, voltage-doubler, bridge-rectifier and provides 180V DC output. Because the remote transformer delivers most of the power to operate the control unit, the undesired magnetic noise of this transformer is isolated from the control unit chassis. The chassis-mounted power transformer operates at a low power level and thus generates a low level of magnetic noise.

The control unit (*Photo 2*) has a two-section, six-position selector switch. Five positions are used for line-level operation, while the sixth position can be used for a phono amplifier. A DPST power switch is used to turn the control unit and remote power amplifier on or off.

The control unit operating in the line-level or phono mode with volume control three-fourths on is "dead quiet," and your ears cannot detect noise inches away from the speakers. The realism of sound reproduction of the control unit with the phono and line-level amplifier operating is excellent. A 100k volume control (P1) with a 40% tap, a 250kW balance control (P2), and a 500kW bass enhancement control (P3) are employed as the basic operating controls.

## LINE AMPLIFIER

The line amplifier (*Fig. 1*) uses a 5687 paralleled SRPP circuit with a voltage gain of 10. The paralleled 5687 SRPP circuit provides an output impedance of 2600W and obviates the need for a cathode follower. The distortion at the output of a paralleled 5687 SRPP driving either a 100kW or a 10kW load is unchanged at 0.18% at 2.5V RMS/1kHz. The heaters are operated at 12V DC that

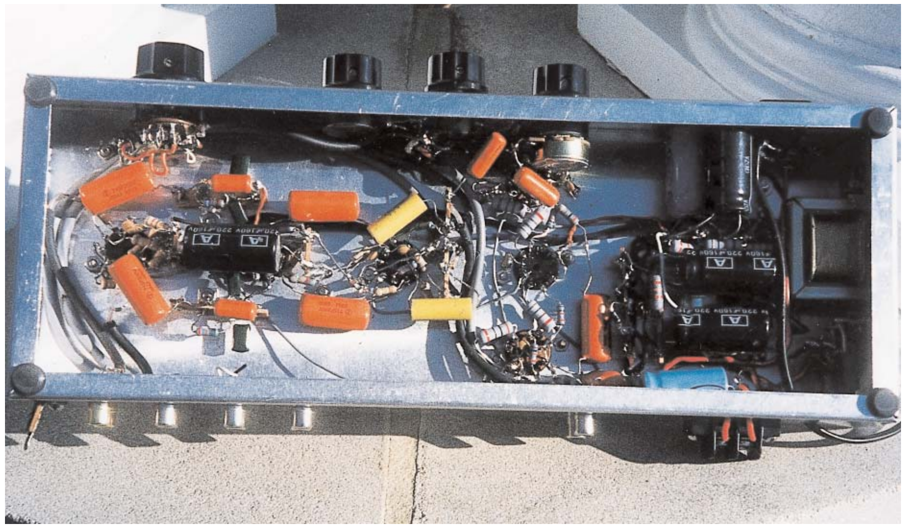


**PHOTO 2: The control unit (internal view).**

is supplied by a filtered, full-wave bridge-rectifier. The plate of the SRPP-connected 5687 is operated from a supply voltage of +180V DC.

The circuit of the paralleled 5687 SRPP amplifier is simple and uses only one coupling capacitor in its output. The 5687 is especially suited for use as a line amplifier with its 3.6V DC cathode bias and its 0.18% distortion at 2.5V RMS output. This provides considerable headroom at both the input and the output of the tube and ensures no clipping will occur during reproduction of music peaks.

The noise of the line amplifier is 0.3mV with volume control (R1) full on and the input open, non-shorted. (The noise measurement is made using a DMM, Radio Shack, 22-168A.) All noise and ripple measurements throughout this article use a DMM with shielded test leads.

The distortion from 30Hz to 15kHz of the line-level stage is typically 1.0% at 15V, 0.64% at 10V, 0.52% at 7.5V, 0.32% at 5V, 0.25% at 3V, and 0.1% at 1V RMS output. Clipping of the paralleled 5687 SRPP circuit occurs at 30V RMS. The 100Hz and 1kHz square waves are reproduced perfectly and the 10kHz square wave shows only a slight amount of "rounding," as is expected with its high-end frequency range of 100kHz.

The normal operating range of the line amplifier is ½ to 3V RMS. The frequency response of the line amplifier is flat at 5V RMS from 20Hz to 40kHz, tapering to 4.2V RMS at 100kHz when feeding a 6¢ audio cable (Radio Shack 15-1505).

**Of interest**: A paralleled 5687 has approximately the same characteristics as the 6H30. You could adapt this circuit for line amplifier use only and eliminate the phono section, which would greatly simplify the construction process.

## BASS ENHANCEMENT CONTROL CIRCUIT

The bass enhancement control circuit (*Fig. 1*) operates by frequency selective capacitor 0.01mF (C1) and negative feedback that is applied between the plate and grid of 5687 (V1). The 0.01mF capacitor blocks all frequencies below 400Hz from entering the negative feedback loop of the 5687. The 0.01mF capacitor is fully functional to all frequencies above 400Hz.

The level of the negative feedback is controlled by 500kW potentiometer (P3), and control does not start until the potentiometer setting is rotated below 300kW. The maximum amount of feedback that can be applied to the 5687 is controlled by a 39kW resistor (R1). Resistor R1 is chosen to provide a typical maximum reduction of the input signal from 1V to 0.5V.

The feedback is applied to the 40% tap of volume control potentiometer (P1) that essentially applies the feedback directly to grid (V2). To observe the feedback effect as feedback control (P3) is set to 250k, 150k, 100k, and 0W, refer to *Fig. 4*. Keeping the output voltages constant at 20Hz makes it easier to evaluate the effects of the inverse feedback at the four potentiometer settings. As the volume control is rotated towards the three-quarter fully open position, the feedback is automatically reduced by the action that occurs at the 40% tap of volume control P1. This reduction in the bass enhancement as the volume level is increased is in accord with the Fletcher-Munson response curve. Also of interest, as the feedback control is increased to a maximum setting, you can observe from the bass enhancement chart (*Fig. 4*) that the slope from 100Hz to 400Hz is greatly increased.

The advantages of this control over the traditional tone controls:

1. Does not require a dedicated tube; uses existing tube. Feedback is employed between grid and plate.
2. Feedback is limited to a typical 6dB; therefore, bass enhancement is 6dB maximum, not 12 to 18dB as in conventional circuits.
3. Bass enhancement is limited to 10 to 100Hz, slope from 200 to 400Hz, and feedback attenuated frequencies are flat from 400Hz to 40kHz. When traditional circuits are used, the frequency alteration extends beyond 2kHz.
4. The application of inverse feedback with this circuit reduces the distortion for all frequencies from 400Hz to 40kHz. In traditional circuits, the distortion is increased.
5. No phase anomalies as in traditional circuits.

The only negative is that operating this circuit requires the use of negative feedback, but these effects are only noticeable as the feedback control is rotated beyond the three-quarter way mark. The feedback control (P3) is inoperative when set above 350kW or fully open position and thus has no effect on the frequency of the line amplifier. This method is simpler and less expensive than installing a switch to disable the feedback function.

## ALL-VACUUM-TUBE PHONO AMPLIFIER

The phono amplifier (*Fig. 2*) uses two 6N1Ps and one 6922. It features a non-feedback design with a passive RIAA network. The noise is less than 2.4mV at the output and sensitivity is 3mV for 0.68V RMS output at 1kHz. The distortion of the first, second, and third stages is less than 0.15% from 20Hz to 20kHz with a 0.4V RMS output in each of these stages. The distortion measurements are made bypassing the RIAA circuit.

The frequency response of V4, V5, and V6 with Inverse RIAA Network connected to the input of V4 is shown from 20Hz to 20kHz (*Fig. 5*). I used an Inverse

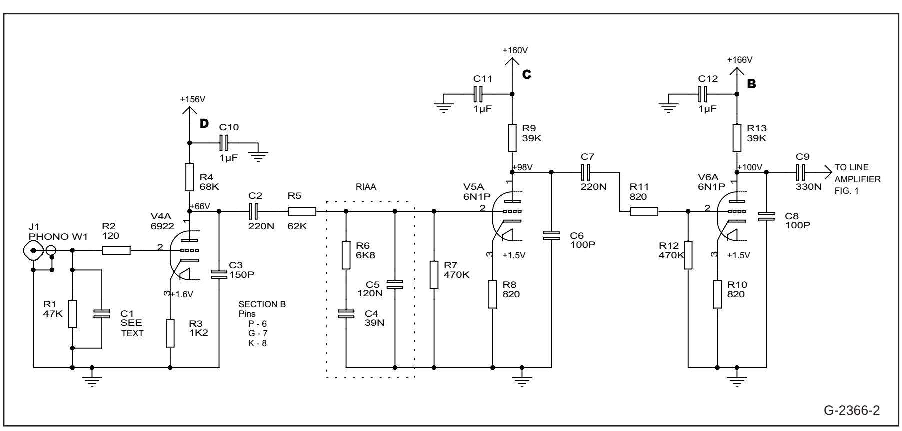



**FIGURE 2: Phono amplifier schematic.**

RIAA Network (Norman L. Koren, 2/97 *GA*) to ensure that the circuit components chosen for the RIAA network provided a flat frequency response. If an inverse RIAA network is not available, the information provided in *Table 2* may be used to determine the accuracy of the phono amplifier RIAA network.

With 3mV, 1kHz fed to the input of V4, the output at V4 should indicate 42mV. The loss through the RIAA network drops the 42mV to 3.6mV at the input of V5. The voltage at the output of V5 is 58mV RMS.

The 58mV output of V5 feeds the third stage V6 and the output of V6 is 0.68V. The output of the third stage V6 feeds the line amplifier (5687). The output of the paralleled 5687 line amplifier is 6V RMS (5687) with 3mV, 1kHz at the input of the phono amplifier. With a record playing, the output of the line amplifier typically swings from 1 to 9V AC.

The noise at the output of the line amplifier with a six position, rotary switch set to the phono position is typically 5mV (with the 600W load of the audio oscillator across the phono input and the audio oscillator turned off, simulating the phono cartridge load). Capacitor C1 tailors the high-frequency output of the magnetic pickup, and R1 provides the proper load to the cartridge.

The value of capacitor C1 is determined from the cartridge-recommended capacitance value. It is obtained by subtracting the cable capacitance, the input circuitry, and amplifying device capacitance from the cartridge-recommended capacitance value. When a range is given for the cartridge-recommended capacitance value, it is suggested the lowest figure be used for the calcula-

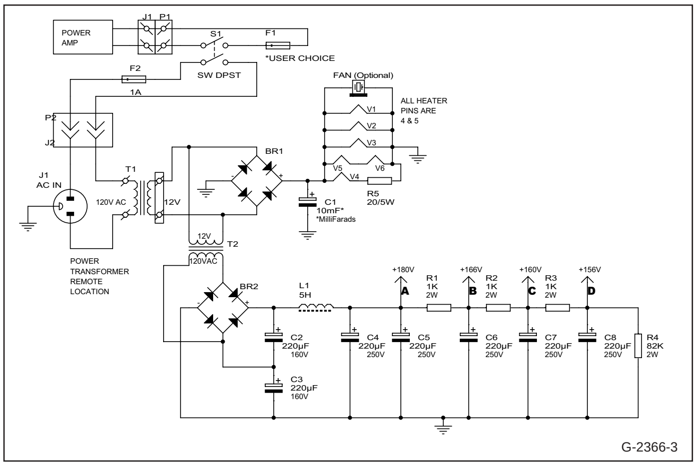



FIGURE 3: Power supply schematic.

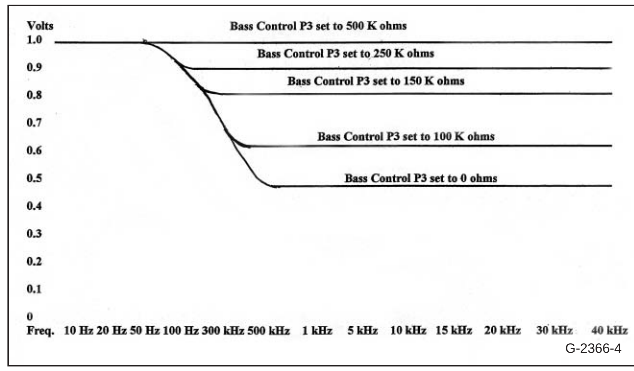



FIGURE 4: Bass enhancement circuit frequency response curve.

tion. A further refinement of the selection of the phono cartridge capacitance is presented in Raymond A. Futrell's article, "LP Terminator" (*audioXpress*, Jan. 2003).

Futrell's "simplified" formula, which assumes the loading resistor is 47kW , Q, 0.707, and the mH value is above 100 (moving magnet (MM) cartridge) is: C (in pF) = L (mH) times 0.225. This simplified formula only applies to MM phono cartridges. C = total value of shunting capacitance paralleling the coil of your cartridge.

Remember, the typical shielded phono cable has a capacity of 25 to 37pF per foot (confirmation requires measurement because some cables have much higher capacity/per ft. values and would be unusable). Also, a 6922 has a typical 30pF of capacitance and a high mu vacuum tube input circuit about 50pF of capacitance. This phono amplifier is designed for MM cartridges having outputs of 3 to 8mV.

## CARTRIDGE CAPACITANCE CALCULATION

A simplified method of tailoring the cartridge for the proper loading capacitance is to use the chart on *Table 1*, which is derived from Futrell's calculations. When the manufacturers of phono cartridges state the value of the recommended loading capacitance, they are including the capacitance of all the input circuitry from the phono cartridge to the input of the 6922 vacuum tube (including the capacitance of the 6922, about 30pF). Also included is the capacitor at the input of the phono amplifier and the capacitance of the phono cable. When this concept is accepted, the manufacturers' recommended value of the loading capacitance becomes acceptable criteria for understanding the capacitance rating system.

Most manufacturer capacitance rating systems appear to be valid or a reasonable compromise when this additional insight into capacitance loading is understood. You must review your system to ensure the total capacitance conforms to the cartridge manufacturer's recommended total loading capacitance value. If your total phono system capacitance matches or is less than the cartridge manufacturer's recommended loading capacitance value (but not less than the value determined from Futrell's L/C formula), you'll obtain good wide frequency sound from your phono cartridge.

A review of the phono amplifier, phono cable, and input circuitry capacitance will in most cases reveal that your system is responsible for the major capacitance imbalance of the phono cartridge capacitance loading. After you have confirmed your system meets the recommended loading capacitance of the manufacturer, but you still desire to improve your system, then proceed to Futrell's (L/C resonance) mathematical formula for further refinement of the cartridge capacitance loading process.

**Important**: If you use the chart on *Table 1* to match your phono cartridge to your existing phono system, the input loading capacitor of the phono amplifier (if there is one) must be disconnected. This phono amplifier did not experience RF interference using 3 to 10¢ lengths of SSPC or coaxial cable.

If you have costly video or audio cables with low capacity values compatible with RG-59, you can use them in this application. The capacity of these costly cables must be measured, as they do not offer capacity values. All audio cable and many phono cartridge manufacturers prefer to withhold detailed specification information.

**Important**: The five steps illustrated in *Table 1*, the manufacturer's recommended load capacitance, or Mr. Futrell's mathematical concept may be used in selecting loading capacitance values.

**Note**: In the pamphlet supplied with the cartridge, the manufacturer does

### TABLE 1
### SUGGESTED CAPACITANCE LOADING, WITH CARTRIDGES HAVING AN INDUCTANCE OF 400 TO 1000MH:

| 400-500mH | 500-600mH | 600-800mH | 800-1000mH |
|---|---|---|---|
| 1. For 3¢ cable length, use: RG-59, C1 is 22pF | SSPC* | SSPC, C1 is 33pF | SSPC, C1 is 82pF |
| 2. For 6¢ cable length, use: RG-62* | RG-62, C1 is 22pF | RG-59* | SSPC^* |
| 3.For 8¢ cable length, use: RG-62#* | RG-62* | RG-62, C1 is 22pF | RG-62, C1 is 68pF |
| 4.For 9¢ cable length, use: N.R. | RG-62#* | RG-62, C1 is 12pF | RG-62, C1 is 56pF |
| 5.For 10¢ cable length, use: N.R. | N.R. | RG-62* | RG-59#* |

**Note**: (The 3¢ cable is the record player cable, except for the 3¢ RG-59 coaxial cable)
**SSPC:** standard shielded phono cable with typical capacitance of 32pF/ft (requires measurement).
**RG-59/U:** coaxial cable with capacitance of 20pF/ft, 75W has RCA male connectors both ends, 6¢ length.
P.N. 32-4545, $1.42 per cable, 10¢ length; P.N. 32-4550, $1.62 per cable, distributor MCM Electronics, 1-800-543-4330.
**RG-62/U:** (13 pF/ft, 0.2†OD, copper braid, Radioshack.com, P.N. 910-1598) is recommended, requires adding RCA male connectors (Radio Shack P.N. 274-321).
Use of coaxial cable requires removing the phono cable attached to the record player and installing two chassis-mounted RCA female connectors (Radio Shack P.N. 274-852) at the rear of the record player.
*Capacitor C1 is not required in this chart, the 6922 input capacitance is assumed to be 30pF. If you use a high mu triode, subtract 20pF from the value of C1 in the chart.
#Exceeds L/C calculations: Note 3 (RG-62) by 25%, Note 4 (RG-62) by 16%, and Note 5 (RG-59) by 12%; all excess values meet manufacturer rating. All L/C calculations are mid-point averages of the mH range values.
^For 6¢ SSPC cable, add a 3¢ SSPC extension cable (Radio Shack P.N. 42-2353, 30pF per ft) to the 3¢ record player cable.
N.R. This length of cable is not recommended for this mH range.

### TABLE 2
### RIAA FREQUENCY RESPONSE OF PHONO AMPLIFIER

If an Inverse RIAA network is not available, the output voltage of the phono/line amplifier is checked against the simulated output of the actual RIAA curve.

| | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|
| RIAA Actual (V RMS) | 0.92 | 0.71 | 0.45 | 0.26 | 0.14 | 0.10 | 0.074 | 0.039 | 0.02 |
| Phono/Line Output | 1.0 | 0.80 | 0.5 | 0.28 | 0.14 | 0.10 | 0.076 | 0.042 | 0.022 |
| Frequency | 20Hz | 50Hz | 100Hz | 200Hz | 500Hz | 1kHz | 2kHz | 5kHz | 10kHz |

**Note**: The input voltage of the phono amplifier is adjusted for a 3mV RMS/1kHz signal and the volume control P1 of the line amplifier is set for an output from the line amplifier of 0.1V RMS. The input signal level and P1 volume control setting are not changed for the remainder of the frequency measurements.

not specify that a given capacitance be added to the cartridge circuit, only that a total recommended amount of capacitance is present.

The cartridge specifications are required for proper integration and mating to the phono amplifier. It is important to know the standard shielded phono cable (SSPC) used in your system is 25 to 35pF per ft. Do not assume it has standard capacity values, it could have very high capacity values. If the phono cable in use is attached to a record player and is 3¢ in length, it probably has standard capacity values.

If, however, your system requires a 6-10¢ cable, make certain that it is a low capacity cable. A high capacity cable could make matching your cartridge with the correct loading capacitance impossible. My chart simplifies the cable selection process.

My phono system uses a 10¢ RG-59U cable connected to a Stanton 680HP phono cartridge (930mH) and is compatible with Futrell's L/C concept and my chart. After making this modification, I noticed a definite improvement in the sound of my audio system, particularly the high frequencies. Also, the 10¢ coaxial cable is free of hum/noise and RF interference.

This phono amplifier has an output of 0.68V RMS at 1kHz with a 3mV input signal, loaded by a 100kW volume control and 250kW balance control. The signal is higher than that of a Sony CA9ES CD player. The volume control is set at the one-quarter to one-third position for living room "listening" levels when in the phono mode. This confirms the high output design of this all-vacuum tube unit.

The description and function of the individual circuit components of the phono amplifier (*Fig. 3*) follow:

- Capacitor C1 (*Table 1*) tailors the high-frequency output of the magnetic pickup, and R1 provides the proper load to the cartridge.
- Resistors R2 and R5 provide resistance de-coupling.
- Capacitors C4, C5 and resistors R5, R6 form the RIAA network.
- Capacitors C2, C7, and C9 are coupling capacitors.
- Capacitor C3 suppresses RF pickup, capacitors C6, C8, C10, C11, and C12 are required to stop unwanted high-frequency oscillation.
- Resistors R4, R9, and R13 are plate load resistors, and R3, R8, and R10

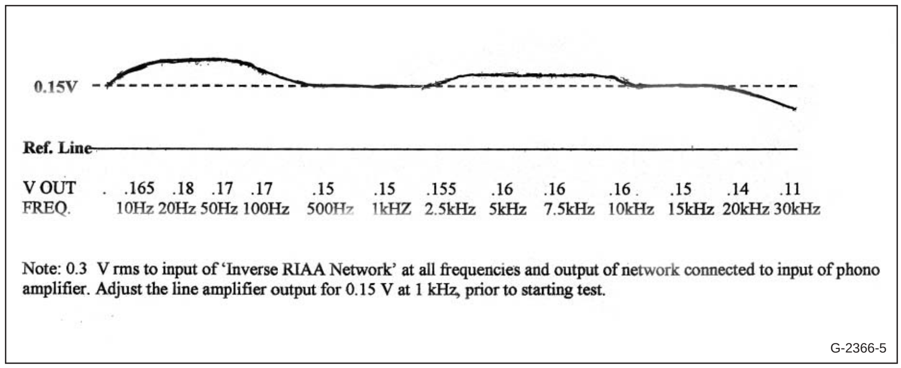


**FIGURE 5: Inverse RIAA frequency response curve of phono amplifier.**

are cathode bias resistors, which were chosen to provide low distortion operation for V4, V5, and V6.

**Note:** You can add a 1000mH, 14W , RF choke (Mouser P.N. 542-77F102) to the input of the phono amplifier, if needed, for RF suppression. Place the choke between the phono jack and 120W resistor.

**Important:** I recommend that you have an audio oscillator (optional) and vacuum tube voltmeter or digital multimeter (DMM). This equipment is essential to ensure proper balance and that gain is obtained throughout the phono amplifier circuit. The 6N1Ps and 6922 provide a very broad frequency response with a detailed and clean output.

## POWER SUPPLY

The power supply (*Fig. 3*) consists of a remote 120V AC to 12V AC transformer T1. The transformer secondary is rated at 8A at 12V AC, which is conservatively rated to furnish the 3.3A required to operate the 6N1Ps, 6922, and 5687s. The remote transformer 12V AC output voltage feeds a full-wave, bridge-rectifier that is filtered by capacitor C1. The output of the heater supply provides 12V DC to the heaters with a ripple of 0.32V RMS. The 6922 heater requires voltage-dropping resistor R5 to obtain 6V DC. The 12V AC output of the remote transformer also feeds the 12V AC winding of chassis-mounted transformer T2.

The 120V AC output of transformer T2 rated at 1.5A feeds the full-wave bridge (D1 through D4), voltage-doubler rectifier. Voltage doubling is accomplished by capacitors C2 and C3. The choke L1 diminishes the switching transients that occur during the rectification process. Capacitors C4 through C8 are additional filtering devices and provide a low-noise ripple of 0.6V RMS at the +180V DC output. (The noise measurement was made with a DMM using shielded test leads.) Resistors R1 through R3 provide de-coupling. The resistor R4 is the bleeder for the 180V DC output. Transformer T2 is only slightly warm after hours of operating time.

**Note**: I originally used a 5A rated transformer for T1 but this transformer became quite hot, so I recommend you use 8A transformer (553-F8-12 at $25) from Mouser Electronics. Safety shielding is required for AC line terminals.

**Forced air-cooling:** Blower fan B1 is optional; if used, locate it at the end of the chassis facing the transformer.

## CONSTRUCTION

**Important**: It is imperative that the control unit (*Photo 2*) is an unpainted aluminum chassis to ensure that all components are properly grounded. Do not use a painted steel chassis!

All the electrolytic capacitors are grounded at the negative band, except those connected in series. All other components are connected to tube sockets. In the phono amplifier section, seven 5-pin terminal strips are required.

It is very important that the assembly be tailored around the implementation of the seven 5-pin and three 2-pin terminal strips. Also required are eight spade lugs. Using the terminal strips provides a very simple step-by-step construction process. It also permits easy changeability of parts. Building the control unit is a fun, easy project using the 5-pin terminal strips, but a "nightmare" using a PC board. The RIAA network components are mounted on a single 5-pin terminal strip.

It is important that all audio runs approximately 3† or more use shielded cable (especially at the input of the phono amplifier) and that the shield is grounded at the input end. It will only be necessary to ground the shielded leads of the phono amplifier at the end nearest the input of the amplifier. The 5-conductor shielded cable that connects to the 6-position rotary switch must be grounded at the input end. Most grounds are made to the center post, 5-pin terminal strip. The use of chassis-mounted solder ground lugs for the cathode and grid resistors and heater grounds are also recommended.

The minimum test equipment required for building the phono amplifier and the control unit is a vacuum tube voltmeter or DMM (an audio signal gen-

TABLE 3
LINE AMPLIFIER PARTS LIST (DOUBLE ALL PARTS, EXCEPT S1, V1, V2, V3, W1, W2, X1, X2, AND X3)

| | | | |
|---|---|---|---|
| C1 | 0.01m F, 250V | Mouser | 140-PF2A103F |
| C2 | 0.22m F, 200V | Mouser | 75-225P200V0.22 |
| C3 | 100pF, 500V, mica | AE | CSM100 |
| P1 | 100k, audio taper, tap 40% | Radio Shack | RS-271-1732 |
| P2 | 250kW , ½W, linear taper, single | AE | RVA-250KL |
| P3 | 500kW , ½W, linear taper, dual | AE | RVA-2X500KL |
| R1 | 39k, 1W, metal film | AE | RE 39 K |
| R2, R3 | 820, 1W, carbon | AE | RB 820 |
| R4 | 200, 2W, metal film | AE | RF 200 |
| R5 | 100, 2W, metal film | AE | RF 100 |
| R6 | 47W , 1W, carbon | AE | RB 47 |
| R7 | 22 0kW , 1W, metal film | AE | RE 220 K |
| J1-J7 | RCA, phono jacks, female | ALEL | RCMJ |
| S1 | Rotary Switch, 2-pole, 6-position | RS | 275-1386 |
| V1-V3 | 5687 (NOS) | AE | 5687 |
| X1-X3 | T-6 ½ tube socket, 9-pins | PE | 055–500 |
| W1 | 5-cond., 22 gauge, shielded | ALEL | 5C-S22 |
| W2 | Single conductor, shielded | RS | 42-2371 |

**IMPORTANT:** These ⅞†diameter, high quality organic tube sockets provide sleeves that grasp the tube pins firmly. They are recommended for use with high gain amplifier circuits. The cost with shield is $2.25.

TABLE 4
PHONO AMPLIFIER PARTS LIST (DOUBLE ALL PARTS, EXCEPT C10, C11, C12, V4, V5, V6, X4, X5, AND X6)

| | | | |
|---|---|---|---|
| C1 | See text | | |
| C2, C7 | 0.22m F, 200V, 5%, poly. | Mouser | 75-225P200V0.22 |
| C3 | 150pF, 500V, mica | AE | C-SM150 |
| C4 | 0.039m F, 100V, 2%, poly. | Mouser | 140-PF2A393F |
| C5 | 0.012m F, 100V, 2%, poly. | Mouser | 140-PF2A123F |
| C6, C8 | 100pF, 500V, mica | AE | C-SM100 |
| C9 | 0.33m F, 200V,5%, poly. | Mouser | 75-225P200V0.33 |
| C10-12 | 14.8, 250V, metal film | RS | 72-1055 |
| R1 | 47k, ½W, carbon | AE | RA-47 K |
| R2 | 120, ½W, carbon | AE | RA-120 |
| R3 | 1.2k, 1W, carbon | AE | RB-1.2K |
| R8, R10 | 820, 1W, carbon | AE | RB-820 |
| R4 | 58k, 1W, metal film | AE | RE-58K |
| R9, R13 | 39k, 1W, metal film | AE | RE-39 K |
| R5 | 62k, 1W, carbon | AE | RB-62 K |
| R6 | 6.8k, ½W, carbon | AE | RA-6.8 K |
| R7, R12 | 470k, ½W, carbon | AE | RA-470 K |
| V4 | 6922 (SOVTEK) | PE | 6922 |
| V5, V6 | 6N1P | AE | 6N1P |
| X4-X6 | T-6 ½ tube socket, 9-pins | PE | 055-500 |

Important: These ⅞†diameter, high quality ceramic tube sockets provide sleeves that grasp the tube pins firmly. They are recommended for use with high gain amplifier circuits. The cost with shield is $2.25.

erator is optional).

## USEFUL HINTS

1. Tubes V4 and V5 use 2-pin terminal strips for ground and B+ connections. Capacitor C8 of the power supply is located at V4, 2-pin terminal strip.
2. The B+ and B- of BR-1 (power supply) is connected to a 2-pin terminal strip. The 12V AC leads are soldered directly to the BR-I diode AC leads.
3. Capacitors C4 through C7 and R1 through R3 are secured to a 5-pin terminal strip. Capacitors C10, C11, and C12 are connected to the junction of 6922 and 6N1P plate resistors and ground.
4. Diodes D1 through D4 and capacitors C2 and C3 are secured to "back-to-back" mounted 5-pin terminal strips.
5. The 12V AC input power, 12V DC tube heaters wires, and capacitor C1 are connected to a 5-pin terminal strip.
6. A 5-pin terminal strip is used at the output of V6 to secure capacitor C9, B+, and plate resistor R13, grounds of shielded cable. ❖

TABLE 5: POWER SUPPLY PARTS LIST

| | | | |
|---|---|---|---|
| T1 | 120V AC to 12V, 8A | Mouser | 553-F8-12 |
| T2 | 120V AC to 12V, 1.5A | RS | 273-1352 |
| L1 | Choke, 5H, 50mA | AE | P-T155H |
| BR-1 | 4A, 50V AC, PIV | RS | 276-1173 |
| D-1 to D-4 | 1A, 600V AC, PIV, 1N4005, 2 packs | RS | 276-1104 |
| R1-R3 | 1k, 2W | AE | RF-1 K |
| R4 | 82k, 2W | AE | RF-82 K |
| R5 | 20, 5W | AE | RQ-20 |
| C1 | 10,000m F, 16V, electrolytic | Mouser | 647-UVR1C103MHA |
| C2, C3 | 220m F, 160V | Mouser | 647-UVR2C221MHA |
| C4-C8 | 220m F, 250V | Mouser | 647-UVR2E221MRA |
| Fuse holder, 3AG | In-line type | RS | 270-739 |
| F1 User Choice | | | |
| F2 | 3AG, 1A, slow-blow | RS | 270-1021 |
| B1 (optional) | 92 · 22mm, 12V -.12A, 25 CFM, Noise 23dB | | |
| | | Mouser | 670-0D922512LB |
| TS | Terminal strips (3-packs) | RS | 274-688 |
| J1, J2 | 2 position, enclosed shell socket | RS | 274-201 |
| P1, P2 | 2 position, enclosed shell socket | RS | 274-202 |
| TB-1 | Barrier strip, 2 positions | RS | 274-658 |
| S1 | Toggle switch, DPST | Mouser | 10DS059SWD-B |
| Chassis Box | Aluminum, 13.5 · 5 · 2 | AE | P-H14444-18 |
| Chassis box cover plate | 13.5 · 5 | AE | |
| Rubber feet | self-stick, rubber | RS | 64-2342 |
| Solder, rosin core 60/40 | | RS | 64-006 |

**Warning**: Dangerous voltages are present, exercise extreme caution when working on the control unit and never leave the control unit upside down when children are present.

### PARTS SUPPLIER

RS = Radio Shack
AES = Antique Electronic Supply 1-480-820-5411
ALEL = All Electronics 1-800-826-5432
Mouser = 1-800-346-6873
Parts Express = 1-800-338-0531

### TEST EQUIPMENT USED

H-P distortion analyzer, 331A
Heathkit sine-wave, square-wave, audio generator, IG-5218
DMM, Radio Shack 22-168A
Oscilloscope, Proteck, Model 6502

# Room Correction, Part 3

Measurements galore in the third part of constructing a digital filter and room correction system. **By Rune Aleksandersen**

**PHOTO 7: Perpetual Technologies P-1A.**

There are two main types of filters to consider when making digital filters: IIR (Infinite Impulse Response) and FIR (Finite Impulse Response) types. FIR filters, maybe the easiest to use at this time, allow arbitrary amplitude and phase responses. There are two main methods of implementing FIR filters: using convolution in the time domain and handling the filtering in the frequency domain. IIR filters are computationally more efficient but not as easy to handle mathematically.

The Signal Wizard uses FIR filters and handles filtering in the time domain. The filter impulse response and the input signal are convolved to form the output signal. Fortunately, the Signal Wizard accepts an arbitrary ampli-

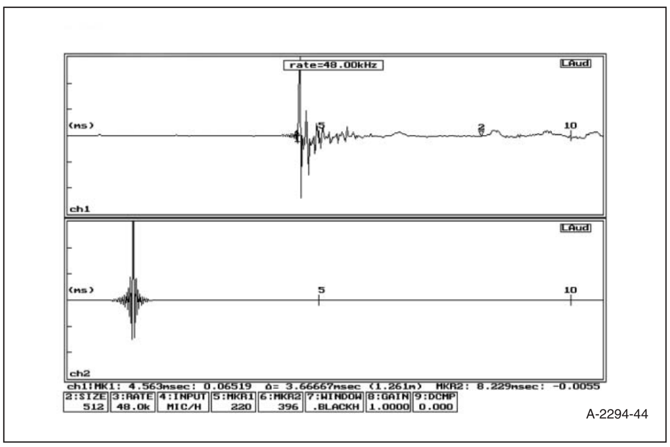



**FIGURE 44: Uncorrected 1m response placing speaker near the back wall.**

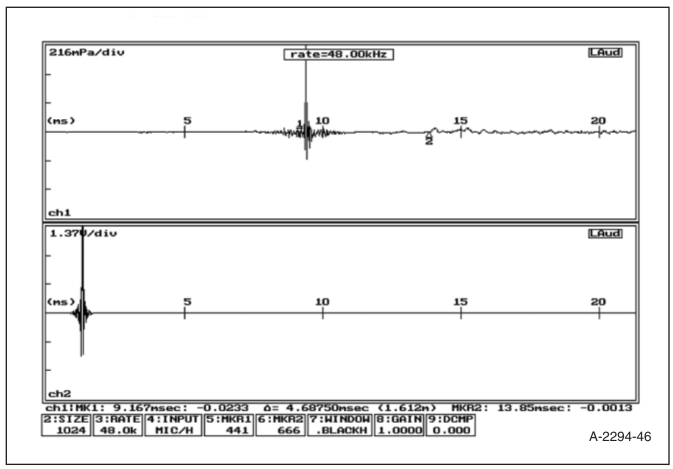



**FIGURE 46: The impulse response after importing the amplitude and phase into Signal Wizard and inverting both.**

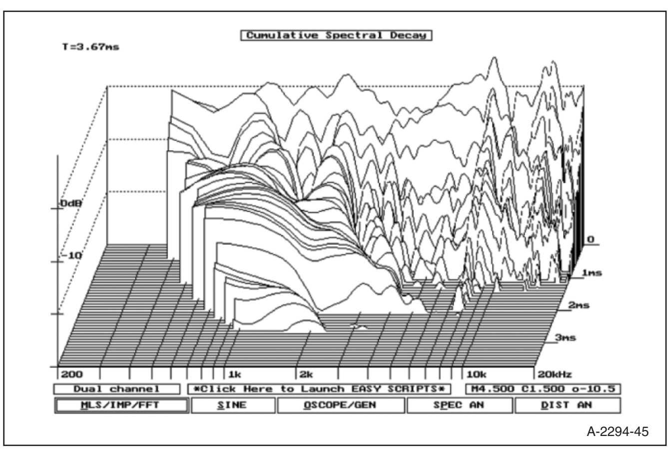



**FIGURE 45: Waterfall of uncorrected 1m response placing speaker near the back wall.**

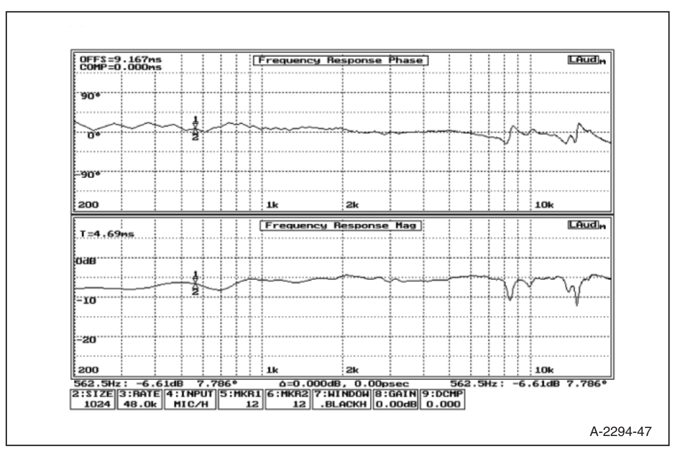



**FIGURE 47: The amplitude and phase of the corrected room response.**

tude and phase response input. From this the Signal Wizard PC software generates an appropriate filter. My task is the most exciting part: to have some Matlab code massage my amplitude and phase responses into my desired

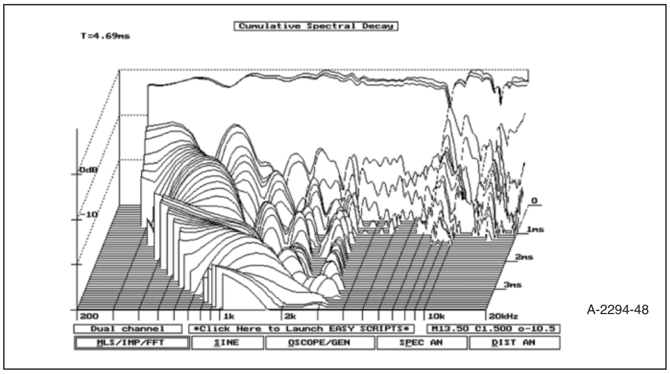



**FIGURE 48: Waterfall of the corrected response.**

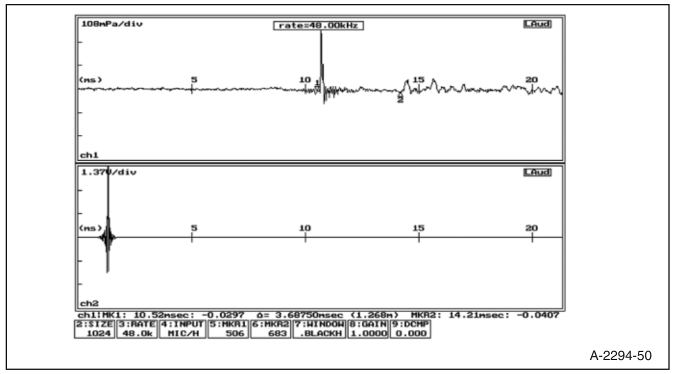



**FIGURE 50: Impulse response of filtered speaker 1m off-axis.**

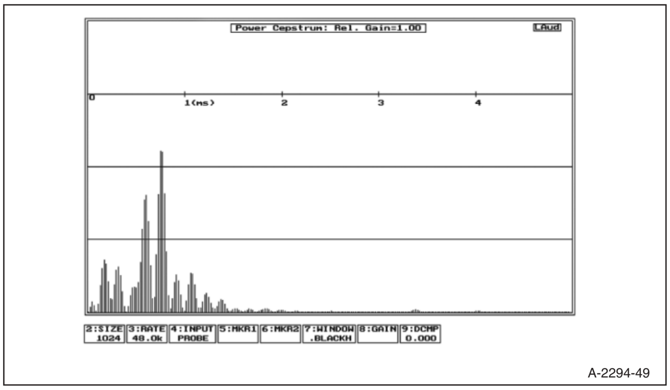



**FIGURE 49: Cepstrum of 1m response with filter.**

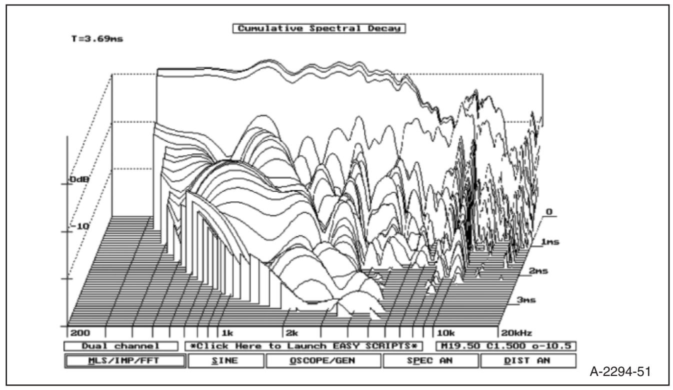



**FIGURE 51: Waterfall response of filtered speaker 1m off-axis.**

filter responses.

The measurements and correction procedures were made of an active closed-box speaker featuring the Jordan JX92 driver. I used LAUD to collect data, and used Matlab to transfer the measuring data into correction curves. I then imported correction curves into the Signal Wizard board and measured the corrected response again with LAUD.

Besides measuring the filters that I am making, I will also do much listening. I am very excited to learn more about the effect of phase distortion audibility[17]. Back in 1990, Greenfield and Hawksford performed some listening tests suggesting a phase correct system gives subtle improvements modifying the sound stage, giving a broader image.

## DIRECT SOUND CORRECTION

Data is collected using impulse measuring techniques. What I measure is a combination of the early reflections and the direct sound of the loudspeaker.

I found that I needed to measure rather closely to mask out listening-position-dependent effects such as floor and ceiling reflections. A distance of about 1.5m and measuring window of around 3.5ms gets me into the far field of the speaker, captures the back wall response, and allows me to reject the floor and wall reflections

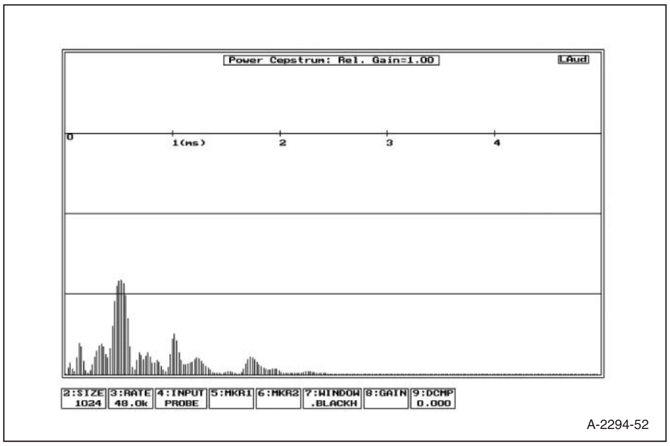



**FIGURE 52: Cepstrum of 1m off-axis response with filter.**

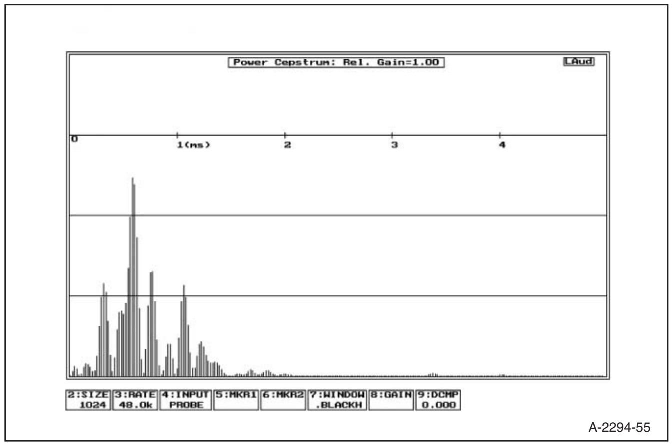



**FIGURE 55: Cepstrum of 1.6m response with filter.**

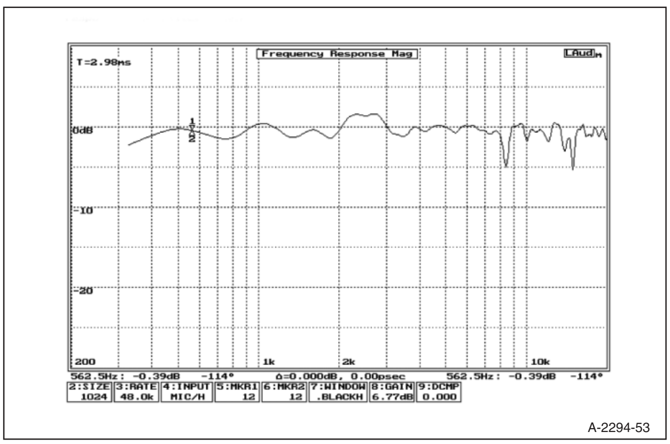



**FIGURE 53: Amplitude response of filtered speaker 1.6m.**

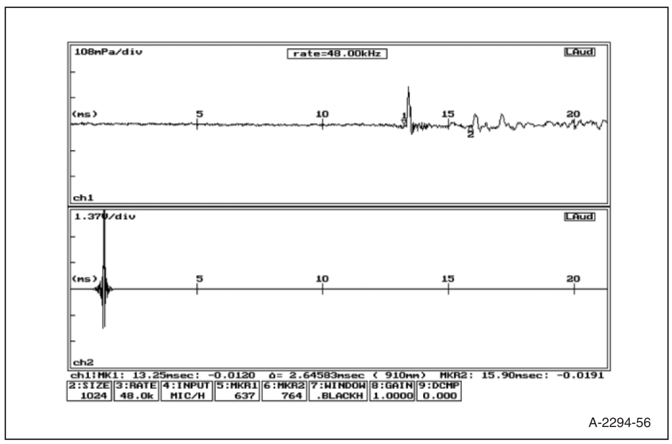



**FIGURE 56: Impulse response of filtered speaker 1.8m off-axis.**

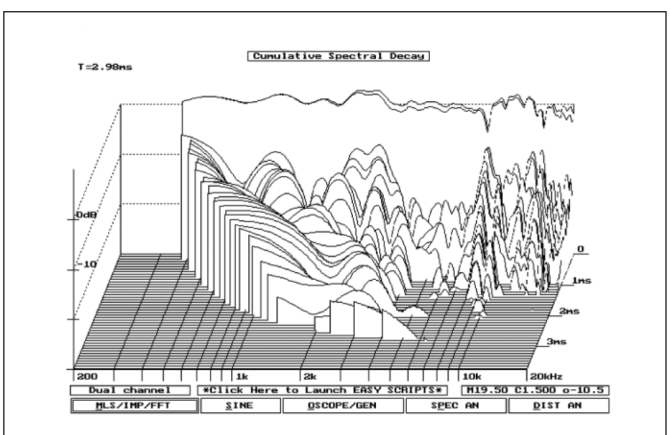



**FIGURE 54: Waterfall response of filtered speaker 1.6m.**

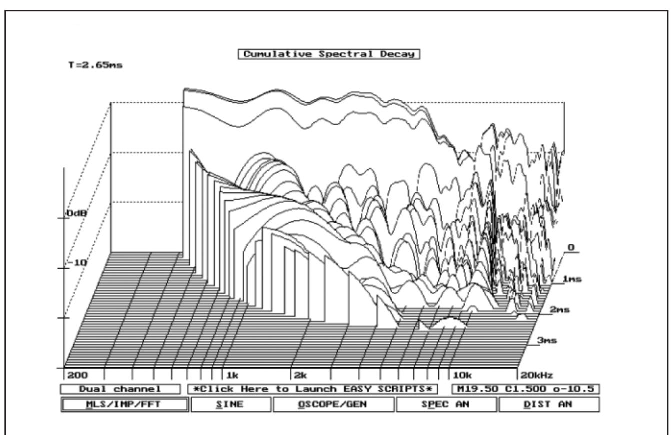



**FIGURE 57: Waterfall response of filtered speaker 1.8m off-axis.**

since those will be outside my measuring window.

In this project, I wanted to try out different algorithms for correcting the direct sound both by measuring and listening. Here are some filter types that I wanted to investigate:

- Time correcting filter—corrects both amplitude and phase flat
- Linear phase filter—correct amplitude flat only
- Psychoacoustics types of correction

### TIME CORRECT FILTER

I used the following procedure for making the time correct filter:

- Measure the impulse response with LAUD 1.5m on-axis
- Import the ⅙ octave smoothed amplitude and phase data from LAUD into Matlab
- Invert the amplitude
- Invert the phase
- Export the data to Signal Wizard
- Measure the results with LAUD at different listener positions.

The resulting impulse response with the Signal Wizard 511 tap filter running at 48kHz is shown in *Fig. 48*. Note the pre-effects before the actual impulse—that is a side effect coming from the digital filter. A filter length of 511 taps at 48kHz equals a filter window before and after the impulse response with total duration of 5.3ms.

For all listener positions, the obtained results are big improvements

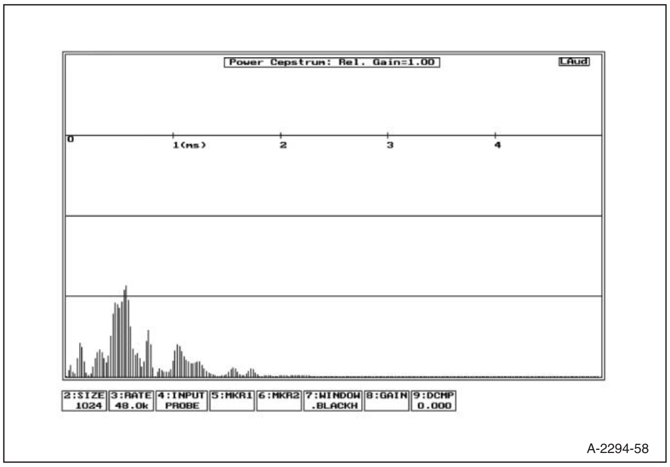



**FIGURE 58: Cepstrum of 1.8m off-axis response with filter.**

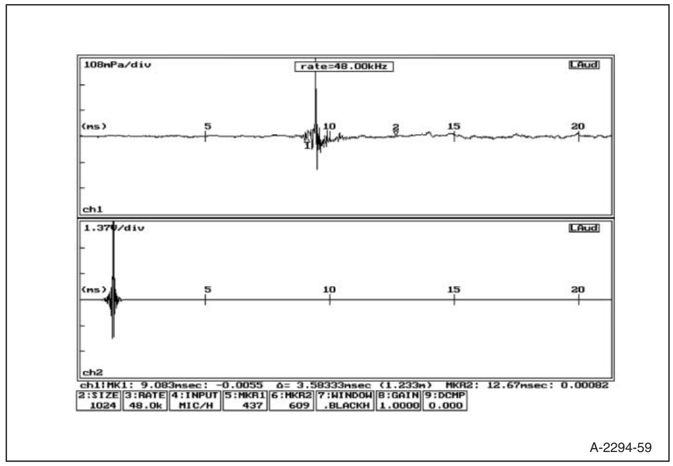



**FIGURE 59: Linear phase filter.**

over the original measured responses (*Figs. 46–58*). On-axis, the back wall reflection and diffraction effects are cancelled out almost totally. You can see this in *Fig. 47*, where there is no visible comb filtering occurring in the frequency response plot. The waterfall plot (*Fig. 48*) shows a very steep falloff for most frequencies, around –20dB within a fraction of a millisecond.

For the off-axis positions, there is some comb filtering present (*Fig. 51* and *Fig. 57*). This is probably due to two factors: the back wall reflection is not totally cancelled out, and the diffraction patterns of the box are different.

What cannot be corrected completely are the driver fundamental resonances. Two of those are predominant: one around 8kHz and one around 16kHz.

Cepstrum responses of all filters show a big improvement over the original responses. Off-axis cepstrum measurements show less reflective energy than the on-axis measurements (*Figs. 52* and *58*). I guess this is caused by the

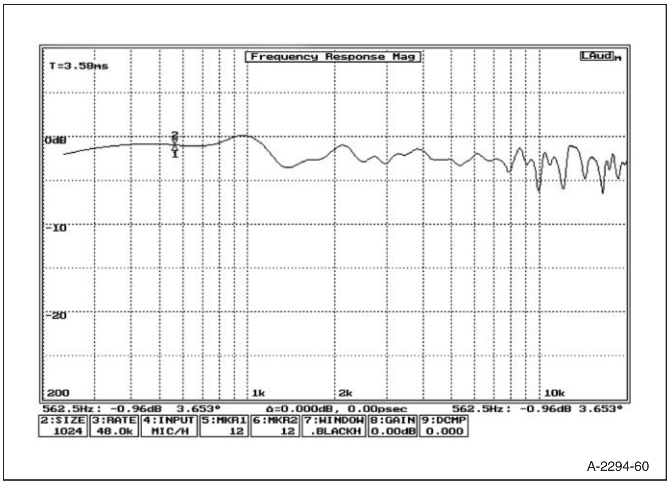



FIGURE 60: Frequency response of linear phase filter.

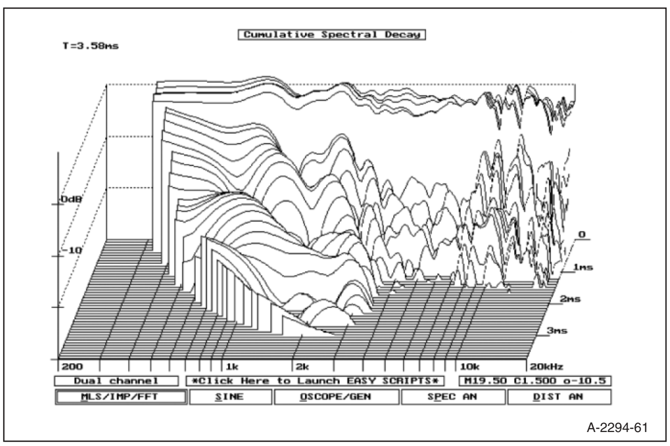



FIGURE 61: Waterfall response of linear phase filter.

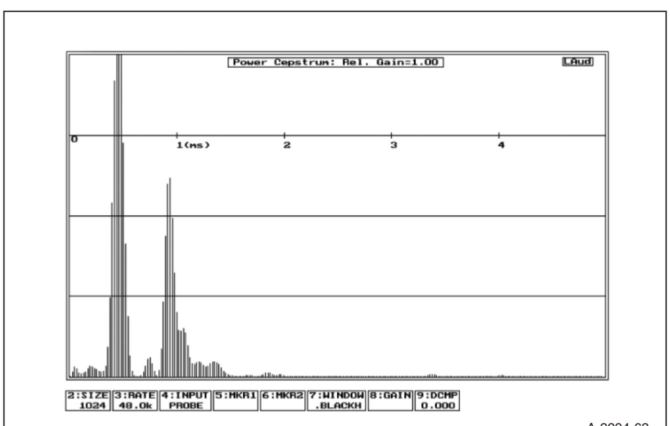



FIGURE 62: Cepstrum of 1m response with linear phase filter.

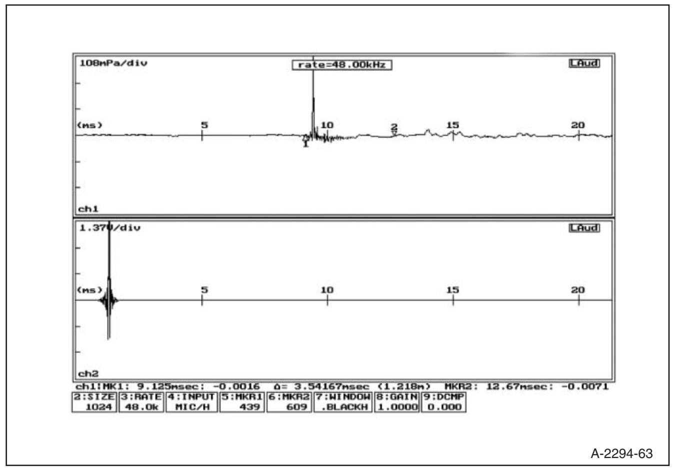



FIGURE 63: Impulse response of filter limiting correcting of dips and peaks to 2dB.

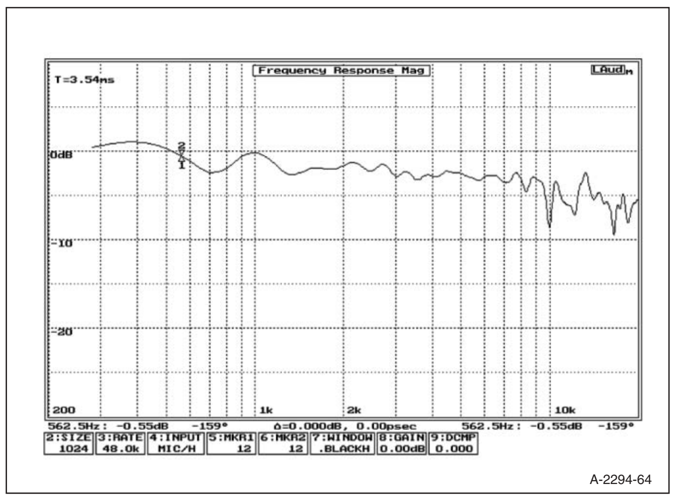



FIGURE 64: Frequency response of filter limiting correcting of dips and peaks to 2dB.

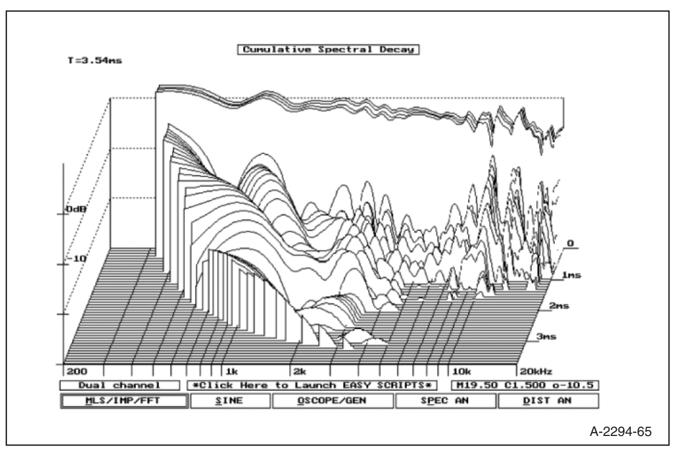



FIGURE 65: Waterfall response of filter limiting correcting of dips and peaks to 2dB.

off-axis response to falloff much above 4–5kHz.

### LINEAR PHASE FILTER

To check out the linear phase filter, the amplitude is inverted while the phase response is set to 0° for all frequencies. *Figure 59* shows the impulse response of the linear phase filter. The impulse response gets cleaned up considerably, but looks different from the phase correcting filter having this S-shape around the main impulse.

Frequency response (*Fig. 60*) looks good, and so does the waterfall plot (*Fig. 61*). The main difference seems to be that the waterfall looks wiggled on the top. And the energy at the lowest frequencies in the plot does not drop as fast as the true phase filter.

A cepstrum plot for the linear phase filter (*Fig. 62*) compared to the cepstrum of the time-correcting filter (*Fig. 49*) indicates some effects from this filter that might not be pleasant to the ears.

### OTHER TYPES OF DIRECT SOUND CORRECTION

Correcting sharp dips at one listener position might lead to nasty peaks in other listening positions. Therefore, it makes sense not to correct big peaks and dips in the amplitude completely. Limiting the correction by amplitude is here shown with a 2dB limit (*Figs. 63, 64,* and *65*). The limiting was first implemented by octave smoothing the amplitude. Subtracting the original amplitude from the octave-smoothed amplitude makes it easy to determine whether the peaks and dips are offending to the ear. I applied a limiting filter before adding this signal with the octave-smoothed signal.

To avoid ill effects from limiting, I used 16th octave smoothing. I applied Hilbert transform last to derive the minimum phase response. The cepstrum in *Fig. 66* is actually a little better than the time correcting filter. This might be caused by the time-correcting filter trying to fix all response problems, including resonance problems that are present in the driver. A little easier going correction strategy might then give a result that is nicer to the ear.

I briefly tried out an averaging filter. The on-axis response was given a weight of 2 while the off-axis response earned a weight of 1. Driver resonance around 16kHz becomes excited (*Fig. 67*), and I think this is why the cepstrum (*Fig. 68*) becomes more dense. Off-axis filtering (*Figs. 69* and *70*) looks no better than when using only the on-axis response for generating the filter (*Figs. 51* and *52*).

### LISTENING TO DIRECT SOUND FILTERS

The uncorrected speaker sounds very good featuring the Jordan JX92 driver. Most notable defects are a somewhat bright and a little edgy sound caused by peaking in the high-frequency area.

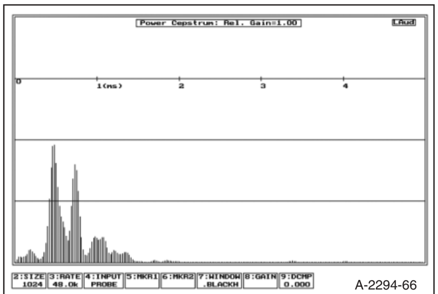



**FIGURE 66: Cepstrum of 1m response with filter limiting correction of irregularities to 2dB.**

Common to almost all filters I tried was that the sound becomes much cleaner. This is probably because the filters flatten the amplitude response.

Both the linear phase filter and time-correcting filter give more definition and easier separation of instruments. But there seems to be a subtle difference between the two: The linear phase filter sounds a little weaker and thinner. The sound could be described as somewhat dirtier or unnatural. Piano (Brahms Piano Pieces, Two Rhapsodies and Fantasies, Naxos 8.550353) sounded a little muddled. I played some organ music (Bach, Toccaten & Fugen Archive 410 999-2), and it is notable that the harmonics line up better.

The image size is the most noticeable difference as the width and height are much larger for the time-correct filter. The feeling of depth and placement of instruments is also bet-

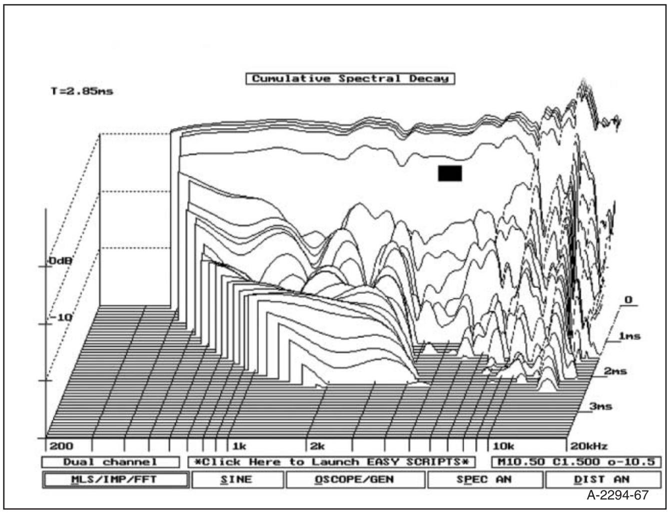



FIGURE 67: Averaging filter measured on-axis.

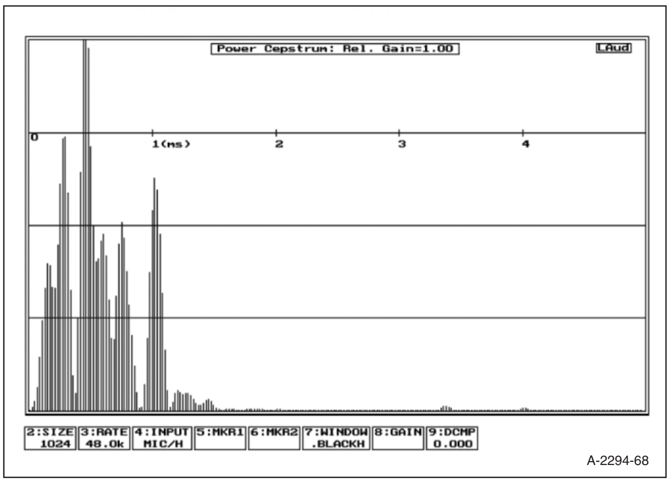



FIGURE 68: Cepstrum of averaging filter measured on-axis.

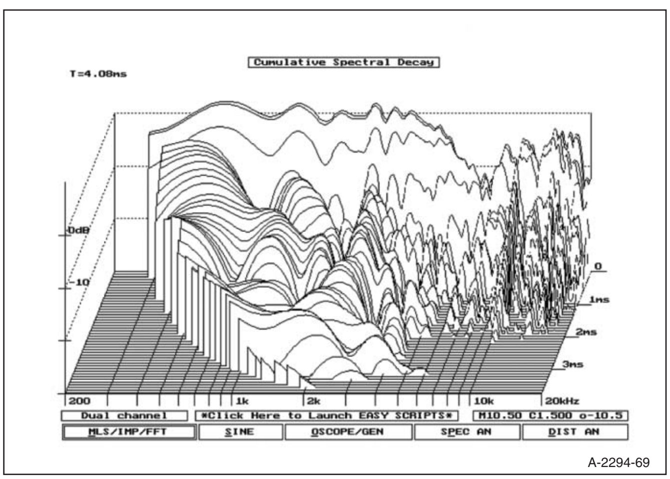



FIGURE 69: Averaging filter measured off-axis.

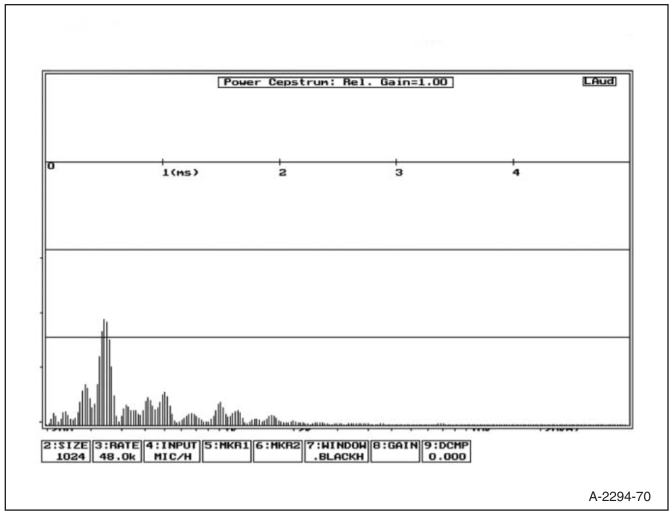



FIGURE 70: Cepstrum of averaging filter measured off-axis.

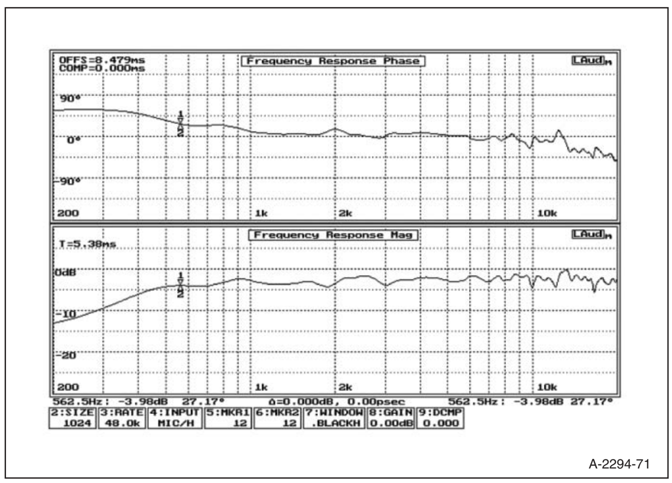



FIGURE 71: Applying upward tilt to curve, minimum phase response.

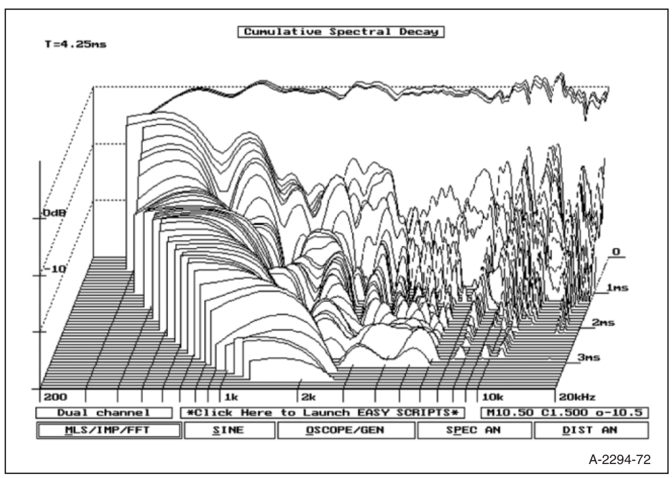



FIGURE 72: Waterfall when applying upward tilt curve.

ter with the time-correct filter than for the linear filter.

The difference in correction to 2dB from the time-correct filter is mainly that the 2dB correction sounds edgier. I guess the reason is probably because this filter does not correct to a perfectly flat amplitude. This edginess is especially pronounced on cymbal and snare drum.

The correction filter sometimes gives the final frequency response a tilt upward or downward, making the balance of the speaker different between filters. Even small amounts of

PHOTO 8: TacT.


PHOTO 9: SigTech.

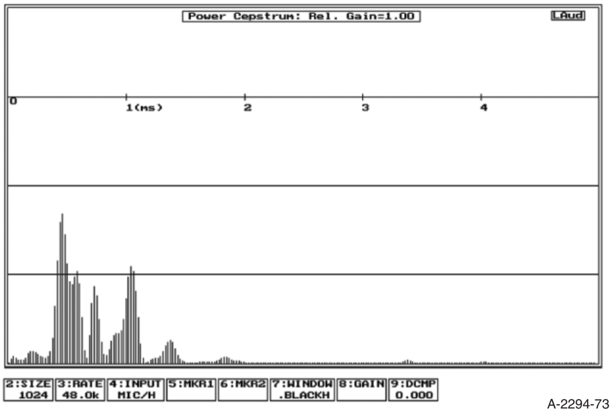



FIGURE 73: Cepstrum when applying upward tilt curve.

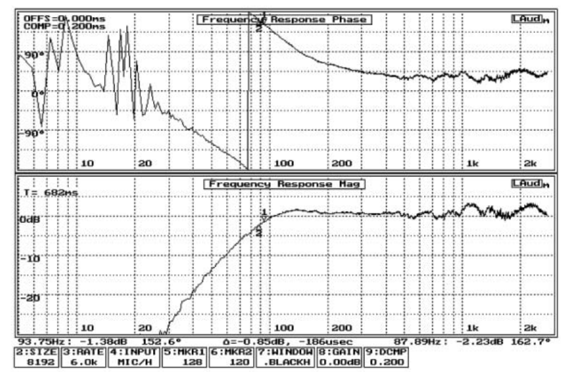



FIGURE 74: Near-field response of JX92 driver.

frequency tilt over wide bandwidths are easily audible. The challenge then is not to be carried away by the different frequency responses and instead try to find the audible differences a correction can make.

**PHOTO 10: DEQX.**

### DIFFUSE SOUND CORRECTION

For diffuse sound correction, I have chosen not to try to do anything with the reverberant sound but concentrate on tilt curves. Spectral balance can easily be adjusted, achieving the spectral balance you desire.

When correcting for direct sound, in practice I sometimes end up with some small tilt in the amplitude response (*Figs. 64* and *65*). This tilt makes the loudspeaker sound brighter or darker than desired.

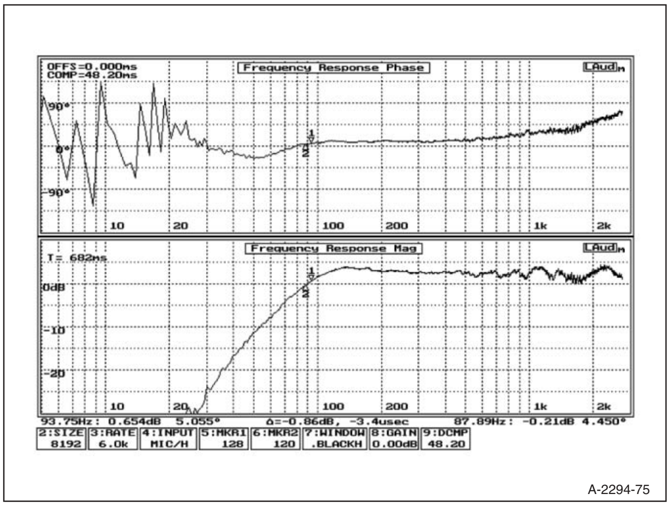



**FIGURE 75: Correcting the phase response of the JX92 driver.**

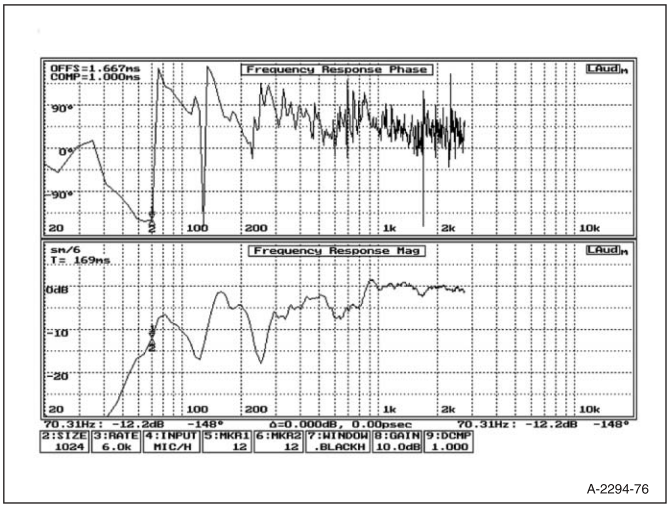



**FIGURE 76: Room response of JX92 studio monitor.**

## COMMERCIAL LOUDSPEAKER AND ROOM CORRECTION SYSTEMS

### PERPETUAL TECHNOLOGIES

www.av123.com

The correction algorithm uses some kind of averaging, measuring on-axis and off-axis both horizontally and vertically. Both phase and amplitude of loudspeaker are corrected. The website contains no details of room correction methods.

(It seems as though something has happened to this company. The room correction option has long been advertised, but not released. The Perpetual Technologies Corporation no longer seems to exist, with its product (*Photo 7*) transferred to AV123, where Mark L. Schifter is still the boss.)

### TACT

www.tactaudio.com

TacT (*Photo 8*) split the band into high-frequency and low-frequency correction bands. They correct both for the room and loudspeaker. Since TacT does not believe in flat correction curves being optimum for the listener, the correction software implements target curves.

### SIGTECH

www.sofgry.com/SigTech/

SigTech use a segmented filter approach, which corrects the loudspeaker for the whole range (*Photo 9*). In their scientific paper, they correct peaks only, since they claim dips in the frequency response are non-minimum phase. Resolution of the SigTech filter is 0.89*Fs/N. When selecting a position to measure as a basis for the correction filter, they select a location with the least amount of nulls (comb-filter effects.) SigTech uses target curves and waterfall plots to examine corrective effects. The SigTech device is beginning to age, and might no longer be available.

### DEQX (CLARITYEQ)

www.deqx.com

DEQX (ClarityEQ) is a new Australian company. They divide the correction into two steps:

1. High and mid frequencies correct semi-anechoic loudspeaker response.
2. Low frequencies (sub 200) correct room response.

ClarityEQ also implements digital brick wall loudspeaker filtering in their DSP units (*Photo 10*).

### SNELL

www.snellacoustics.com/RCS1000.htm

Snells' processor (*Photo 11*) has six channels and a resolution of 1.5Hz featuring 3000 taps. Target curves are implemented as well as smoothing functions. Averaging of up to nine measurements are supported. The Snell unit is no longer in production.

### ULTRA-CURVE PRO

www.behringer.com

Behringer has an interesting unit (*Photo 12*) that measures the room and automatically makes a correction curve. This unit also has some parametric equalizers that could be used for low-frequency compensation. The price is astonishingly low. I use this device for room correction.

### PETER LYNGDORF

www.geocities.com/audiomex2000/History.txt

This is an interesting page posted to a forum by Peter Lyngdorf, the founder of TacT Audio. Here he talks about connections between some of the commercial room correction systems.

Basically, I add the tilting amplitude curve to the direct sound correction curve. Then, I use Hilbert transform to find the phase, the same procedure as when making a direct sound filter. The resulting response is shown in *Figs. 71* and *72*. The cepstrum is shown in *Fig 73*.

Listening to the direct sound filter with and without the tilt curve, you can adjust the spectral balance of the loudspeaker with no impact on the stereo image.

## LOW-FREQUENCY CORRECTION

FIR filters are linearly spaced in the frequency domain, not logarithmic like the way our ears work. This means that high frequencies have a very high resolution, and lower frequencies easily get a too-low resolution. If sufficient resolution is to be gained in the low-frequency region, the filter coefficients will be too many for a DSP processor to cope with.

I set up the Signal Wizard to work with the lowest sample rate common to LAUD. At 6kHz sample rate and 512 defined frequencies, I obtained a resolution of 6Hz.

Data might be smoothed and treated according to psychoacoustics rules. There are excellent Matlab functions developed at the Aalborg University for smoothing according to such rules[18].

I would like to try out the following correction methods:

- Correcting phase woofer transition separately
- Zero phase amplitude correction
- Minimum phase amplitude correction
- Averaging amplitude on different listening positions

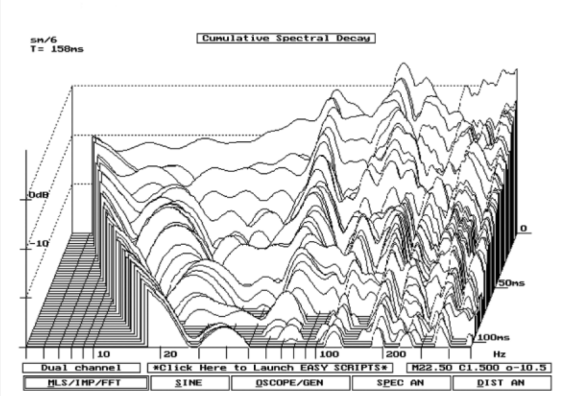



FIGURE 77: Waterfall response of room response.

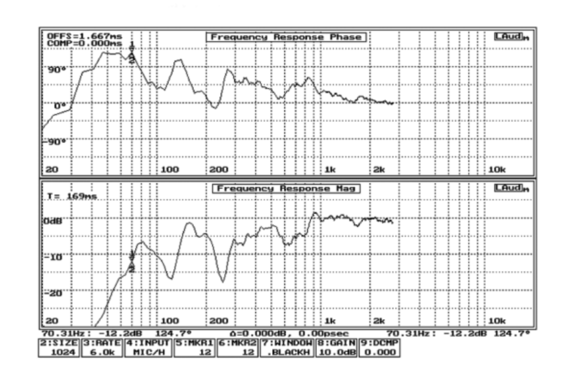



FIGURE 78: Same room position, but Hilbert transform used to derive phase response.

- Correction of audible dips and peaks
- Limiting dips and peaks to avoid filter ringing.

Correcting phase woofer transition separately is easy using an FIR filter. A gradually changing phase of the filter will compensate for the phase turning of the driver and box combination. *Figure 74* shows the near-field response of the JX92 driver mounted in a closed box configuration. The phase turn from DC to around

**PHOTO 11: Snell.**


**PHOTO 12: Beringer.**

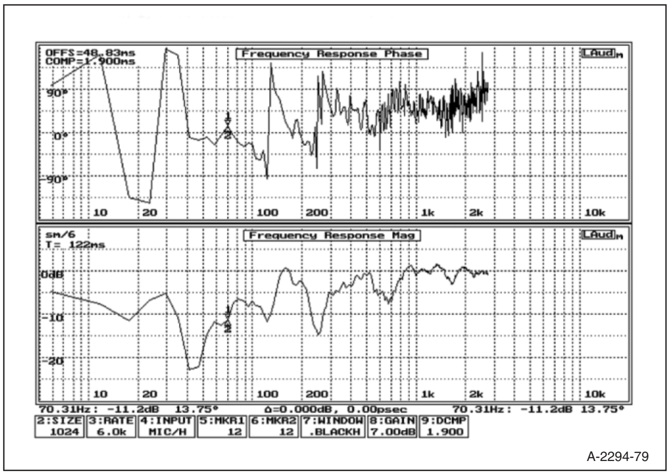



**FIGURE 79: Correcting phase turn effect on room response.**

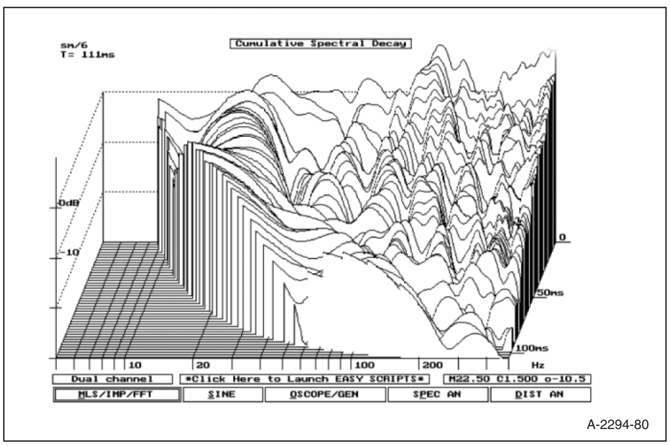



**FIGURE 80: Waterfall response of phase turn filter.**

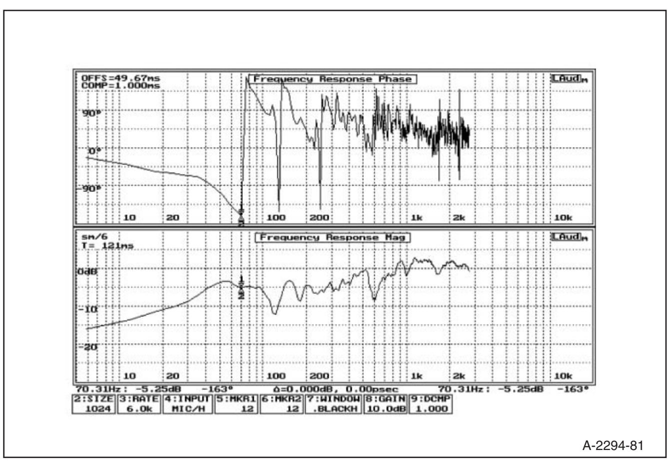



**FIGURE 81: Room response of linear phase filter.**

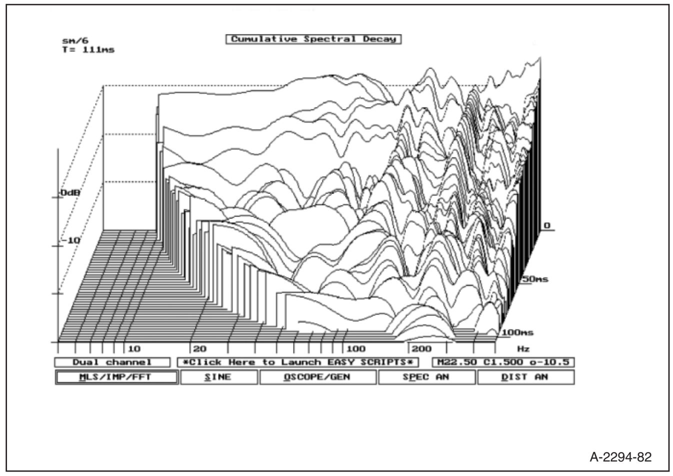



**FIGURE 82: Waterfall response of linear phase room filter.**

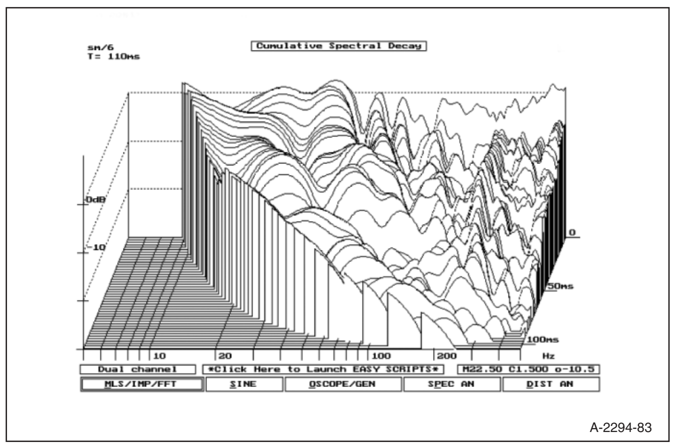



**FIGURE 83: Waterfall response of Hilbert room filter.**

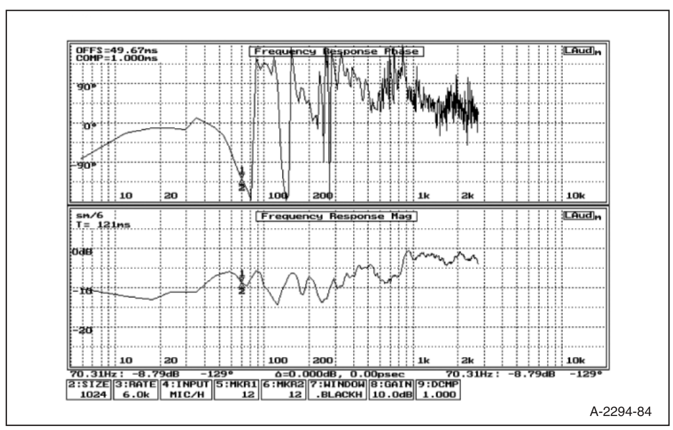



**FIGURE 84: Response of another Hilbert filter.**

400Hz is 360°. (Not as accurate readings below 30Hz are due to the little energy coming from the small driver at those frequencies.) *Figure 75* shows the near-field response using a filter correcting the phase and leaving the amplitude unchanged.

Taking a look at the room response, you can see that the response looks quite uneven for low frequencies (*Figs. 76* and *77*). There are some severe notches introduced by the room, and those notches come sprinkled with sharp notches in the phase response, too. (Note that only the amplitude has been smoothed with $\frac{1}{6}$th octave filters. LAUD does not smooth phase response.) The strange phase leaps suggest those room modes are not minimum phase functions. This suspicion seems correct when you compare a plot of the minimum phase function with the amplitude response measured (*Fig. 78*).

I tried out the phase turn filter and measured the room response (*Fig. 79*). From 100Hz down to 40Hz, the phase response is now much flatter, although there is still the same phase turning from 100–200Hz. The very low frequencies now seem amplified. The waterfall plot is shown in *Fig. 80*.

I tried a linear phase filter to see how it works. The amplitude response (*Fig. 81*) looks like a big improvement, and so does the waterfall plot (*Fig. 82*).

From the linear phase variant, I was curious how adjustments of the phase response would affect the response. I made filters that used the Hilbert transform for deriving the minimum phase response and corrected for this function. I used both the built-in Hilbert function of LAUD (*Fig. 83*), as well as the Hilbert transform in Matlab (*Figs. 84* and *85*). Both filters seem to fall off slower for frequencies below 200Hz compared to the linear phase filter.

I tried a combined filter (*Figs. 86* and *87*), combining the phase turning filter with the linear phase filter. The falloff of the waterfall response seems to be somewhat in between the phase turning filter and the linear phase filter.

From the measurements I made, I do not have a clear sense of which filters will sound the best. The phase turning filter also is an interesting filter. Listening tests will, I hope, reveal what method works the best.

**LISTENING TO LOW-FREQUENCY FILTERS**

The Signal Wizard is not very well suited for correcting the whole frequency range; the filter length available is too short for the filter to have any useful resolution for the low frequencies. Instead of testing with live DSP, I opted for making pre-filtered wave files. I used Matlab to make the filters, and I tried to use it for convolving the files. Matlab handles this badly, and was too slow to be useful. Instead I used the Aurora convolution plug-in for Cool Edit for doing the actual convolution.

The Aurora plug-in is a fantastic tool, making it possible to listen to correction filters on any stereo system without having a DSP processor available. It is even capable of real-time filtering!

I made two types of filters: one linear phase filter and one minimum phase filter. Cutoff frequency was specified to 200Hz, filtering only being active below this frequency where room modes dominate (*Fig. 88*).

It was easy to hear that the linear phase exhibits a more even frequency response for the low frequencies at the listener position than the unfiltered loudspeaker. Compared to the minimum phase filter, the bass of this filter sounds thinner or drier.

The minimum phase filter sounds more natural than the linear phase type. In some recordings this was not easy to hear. Most revealing was organ (Bach, Toccaten & Fugen Archive 410 999-2) and percussion recordings (The Kromata Percussion Ensemble BIS CD-232). Organ pipes probably have a solid overtone structure. The minimum phase filter just sounds more "in-phase" than the linear phase filter, which actually makes the low notes of the organ sound out of tune. Percussion loses timing with the linear phase filter; bass drums sound smeared and unnatural compared to the Hilbert filter.

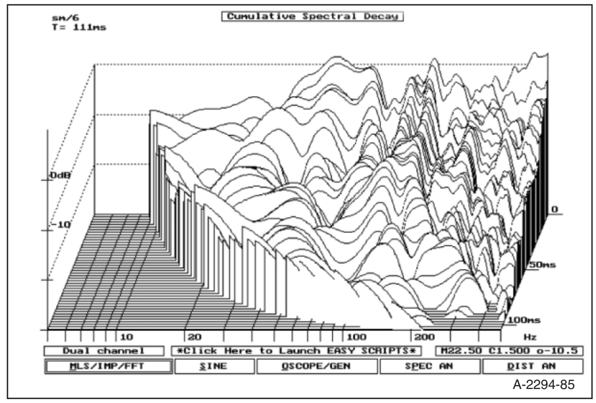



**FIGURE 85: Waterfall response of another Hilbert filter.**

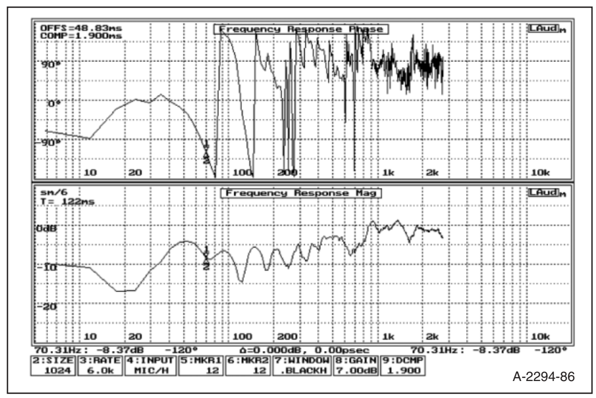



**FIGURE 86: Response of a combined filter.**

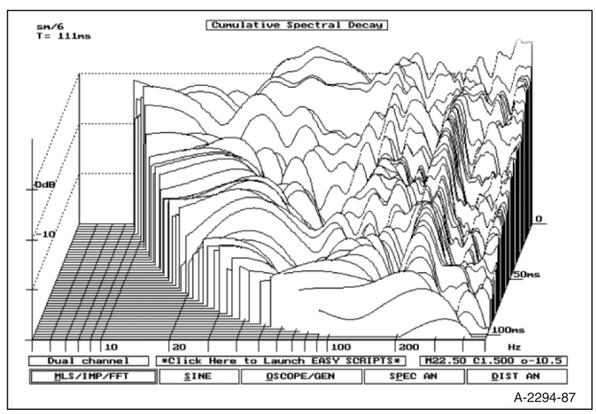



**FIGURE 87: Waterfall response of a combined filter.**

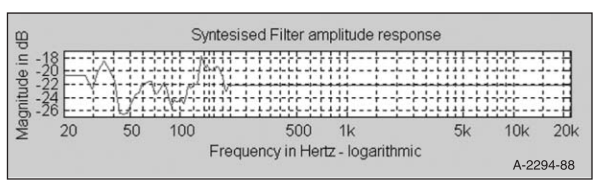



**FIGURE 88: Room filter synthesis.**

## CONCLUSION

It makes sense to divide correction in two main parts:

- Direct and diffuse sound correction
- Low-frequency correction.

Direct and diffuse sound correction can fix loudspeaker defects, including some of the early reflections using rather short filters. Ceiling and floor reflections are much more difficult to handle, and should preferably not be corrected by DSP since those reflections are very dependent on listener position.

Measurements for making the filter should include the ceiling and floor reflections. In an ordinary room, long measurements are not recommended. I was successful using a measuring distance of 1.5m and 3.5ms time windows.

A loudspeaker placed very close to the back wall can produce good results. Both amplitude and phase are important when making a correction filter. Smoothing of the measured data and inverting both amplitude and phase results in a good correction curve. Tilt curves for adjustment of the spectral balance work well for adapting the speaker to the room.

Room correction seems only to make sense for the low frequencies below the Schroeder frequency (100–200Hz.) A minimum phase approach works well for correcting in this band. You should be more concerned with correcting peaks than with correcting dips. Dip correction might turn up as peaks in other places in the room.

For low-frequency correction, you can successfully use parametric equalizers, since these implement minimum phase functionality. You can use IIR filters, which require less computational power than FIR filters. Averaging might be beneficial, although I did not perform a listening test of such filters in this project.

Next month we'll wrap up this four-part series with a look at building the preamp. ❖

### REFERENCES


17. Richard Greenfield, Malcolm Hawksford, "The Audibility of Loudspeaker Phase Distortion," AES preprint 2927(J2) presented at the 88th Convention 1990 March, Mountreux.

18. Kent Pedersen, Thomas B. Sivertsen, Marni Tyril, "Digital Loudspeaker Techniques for Loudspeakers and Rooms," Master Thesis, Aalborg University (1998).

## TUBE LORE

I'd like to respond to two letters in June '04 *aX*. Mr. Hansen ("McIntosh MC2100," p. 64) is quite correct in his letter to Mr. Bruce Brown about the 2N3055 and 2N3772. The 2N3772 can usually be used as a substitute for the 2N3055 under most circumstances. As for transistor cross-references (the same holds true for tube cross-references), the lion's share of the indicated substitutes for a given part will not be exactly the same. Most circuits have enough leeway in operating parameters to allow for a fairly wide range of transistors, but up to 20 percent of substitutes will not work properly in a given circuit.

The original transistor type is always the best to use unless unavailable. Just because a cross-reference recommends a particular "universal" transistor for replacement does not mean the original's and replacement's parameters are the same. Check the small print in these substitution guides. They don't guarantee that any substitute will work in your circuit!

Also. . . just because a tube cross-reference says that a particular tube is a substitute, doesn't mean it is a good substitute. There are many listings in SAM's tube cross-reference books which are just plain poor substitutes, period. For example, the 5751 has become a very popular replacement for the 12AX7. While the 5751 is from the 12AX7 family, it is not a direct replacement for the 12AX7. Several parameters differ significantly. While you might like the results from using a 5751, it is still not a direct substitute.

Mr. Jim Carlyle ("Arcing Prevention," p. 66) writes about an arcing problem in his power amplifier that uses 811As. First off, *none* of the parts inside a tube has been machined. They are all stamp-formed sheet metal and welded. The only source of any kind of powdery residue inside of a tube is either the getter or the cathode oxide. There is absolutely *no* metal dust inside a tube. Mica is non-conductive and could shed some very tiny chips but not powder, so is not the source of any arc-causing material.

The most likely cause of his arc is a weak space charge around the cathode. The space charge acts like a reservoir to supply peak currents, *but* the space charge is only replenished at an average rate. If the electron emission from the cathode is insufficient to build up this "space charge," the cathode coating is called upon to supply high peak currents, which may cause permanent injury to the coating and in extreme cases may even cause sputtering or arcing.

Tubes that have been subjected to this kind of weakening will usually test okay on a tube tester because the tester does not draw the kind of peak currents that can cause the arcing or sputtering (i.e., a tube that is susceptible to arcing or sputtering will likely test above minimum on a tube tester, even into the middle of its test range). Power output tubes that are run hard are susceptible to this kind of failure mode. While it is not very common, it still happens often in amplifiers that are driven hard. Arcing in power output tubes only occurs when the tube's emission level is dropping off, usually after hundreds of hours of use.

The "dust" Mr. Carlyle sees is likely powdery residue from the getter which can form from prolonged high heat levels inside the tube. This can be caused by several conditions: high filament voltage, over-driving the tube, excessively frequent high peak currents, high plate dissipation, and/or a low plate load impedance, among others. I have never seen "dust" inside a new tube, but I have seen this powder inside power tubes that were ridden hard. There is no connection between this getter "dust" and arcing.

High peak currents drawn from the cathode because of a lack of sufficient space charge can cause cathode material to be stripped off the cathode as well. This in turn reduces the emission from the cathode leading to a further reduction in space charge, until the tube arcs!

Regarding another possible point, getter material, if deposited on the grid, *can* cause grid emission; however, this would be extremely unusual and unlikely during the operation of a tube. Deposition of getter material would normally occur only during flashing of the tube, not during operation of the tube. Manufacturers were very careful to make sure no getter material was deposited and all tubes were checked at the factory for any indication of this event before shipping.

I have received questions regarding arcing from many high power tube amp owners, some of which were very particular in the care of their amps. One sign that your output tubes are weakening is a much more positive (i.e., less negative) tube bias setting than with new tubes. Also, any single tube that requires less bias voltage to balance it against the rest of the tubes indicates that it is becoming weak and is a candidate for arcing. High power amps with multiple tubes will benefit from a good fan if they don't already have one. Too much heat will wear out a tube just as fast as overdriving it.

Pay more attention to what your tubes are telling you and you can avoid catastrophic failures and big expenses. I noted that Mr. Carlyle needed to replace all eight output tubes to regain proper balance. This indicates that his other tubes were also on the way out. I estimate that by the time he needs to change his grid bias settings by 8-10V from their initial settings that his tubes will begin to show signs of wear. It is hard to be more precise since I don't have his amplifier schematic to look over.

Tube life is a trade-off between less power, longer life, or more power and shorter life. This has always been the case with tubes.

Edwin G. Pettis
Pettis Engineering
pettis_engineering@prodigy.net

*Bruce Brown responds:*

*I guess I need to defer to the engineers concerning transistors. However, let me give you a few examples from Dynaco engineers and Frank Van Alstine* (Audio *by Van Alstine).*

*There are at least seven known versions of the venerable ST120. The earliest versions used RCA 2N4347 output transistors; the next versions used "select" 2N3055H transistors. The H designates a high voltage version, able to withstand 90V on the collector. I have repaired over 40 ST120s in which someone had tried to use 2N3055 outputs in place of the H units. They all came in blown!*

*The reason for this is that the ST120's regulated power supply is set at about 72V. If the regulator failed, 90V was applied to the outputs either way they don't last too long. The later versions used 2N3772s.*

*Previously an* aX *reader wrote that the 3772 wasn't an audio device and questioned my use of it in rebuilding the McIntosh 2100. If Dynaco was wrong, then I am also, but I guess I would be in pretty good company. I have rebuilt over 20 ST120s with 3772s and not a single one has failed. I can say the same about my 2100. It sounds great, runs cool, and is rock stable.*

*Frank Van Alstine wrote a wonderful article in the August 1984 issue of* Audio Basics, *outlining the problems with the ST120 and his solutions to those problems. It is available free on his website. His recommendation for the output replacements, at that time, was the 2N5630. He also commented on "universal" replacement transistors and noted that most don't even come close to the original.*

*I tend to agree, but unfortunately, many transistors used in vintage equipment are obsolete and, unless you get very lucky at a surplus outlet, you are forced to find a substitute that works. The same holds true for many other tubes, so you have two basic choices find something that will work, or relegate the equipment to the display shelf.*

*Jim Carlyle responds:*

*Thank you, Edwin Pettis, for the stimulating letter describing what should be impossible. Having just passed a magnet near one output tube, I can emphatically confirm that I saw many metal shards up to about 0.5mm long follow the magnet around the inside of the glass. Shocking, isn't it?*

*But I remember 1960, when transistor manufacturers were incredulous at how difficult it was to eliminate factory dust, previously ignored. I have, of course, also seen white oxidized getter inside cooked tubes, but have none myself. Incidentally, my other seven 6DZ7 NOS tubes all remained within bias tolerance. I simply replaced all eight as a peace of mind policy, like doing all four tires.*

## POLYSWITCHES

A comment, if I may, arising from James Moriyasu's comprehensive review of the Usher CP8871 loudspeaker (Feb '04 *aX*).

In an aside to the discussion of resistor placement in the body of his review (p. 59), James writes that he "received a call from a friend who had burned out the resistor in the tweeter crossover of both of his Tannoy TD12 loudspeakers." James suggests this burnout may have been due to the resistor being placed ahead of the capacitors (not so) and then continues (and this is the vital bit) ". . . the resistors were tiny, the size of a dime only thinner, and only rated at 10W." This begs the question, were they in fact resistors? The description fits that of a positive temperature coefficient (PTC) fuse or "polyswitch."

Polyswitches are protective devices that look very much like disc ceramic capacitors, even in some cases being the same light brown color. For currents within their normal operating range, polyswitches present very low series resistance, but if the current exceeds a specified "tripping" value, the internal resistance increases rapidly, limiting the current to a safe level. When the signal returns to normal, the resistance quickly returns to a low value, although there is some hysteresis involved so that with repeated overloads the normally low series resistance may eventually build up to the extent that the device needs replacing. This perhaps also suggests that if subjected to severe, frequent, or sustained overload, the device could burn out.

One application of polyswitches is the protection of loudspeaker systems or individual drivers (particularly tweeters) from accidental excessive signal levels. Where specifically intended for speaker protection, polyswitches may be specified by the nominal impedance and rated power handling of the speaker to be protected. Alternatively, they may be specified by their hold/tripping currents, in which case some simple calculations are required to match the currents to the impedance and power handling of the system/driver.

For system protection, the polyswitch would be placed at the input of the crossover network. If confined to tweeter protection, you should place the polyswitch before the tweeter crossover, not after it.

Consider, as an example, a simple first order crossover, a series capacitor. If you place a polyswitch after the capacitor and an overload current caused its internal resistance to increase to, say, ten times the nominal impedance of the tweeter, the capacitor would "see" an impedance about ten times the design value. Consequently, the rolloff frequency would be lowered by a factor of ten. Certainly, the current in the circuit would be limited to a safe non-burnout value, but the presence of even moderate current at frequencies below the tweeter's fundamental resonance could cause mechanical damage to the fragile suspension.

From all this, it follows that replacing a polyswitch with a resistor would a) probably reduce the normal output level of the tweeter, and b) leave the tweeter with little or no protection against excessive signal levels.

If my suspicion that the "resistors" may be polyswitches proves correct, then I do hope my tardiness in writing this has not made it too late to save the tweeters in James' friend's Tannoys!

Richard Higgins
Port Macquarie NSW
Australia

*James Moriyasu responds:*

*I checked with my friend who owns the Tannoys and he reiterated that the distributor recommended he replace the supposed "polyswitch" with 1.5W resistors. I relayed the information in your letter, and he said he would check with the distributor, again, to see whether they recommend he replace the resistors with the original part.*

*I must confess that I'm not at all familiar with "polyswitches," or similar devices. They just haven't been on my "radar screen."*

## CIRCUIT CAUTIONS

While I applaud Michael Kornacker's resourcefulness in augmenting his LM317-based current regulator with an external power transistor (June '04 *aX*, p. 72), I must express some reservations.

Kornacker's remark that "in the end, unfortunately, you'll need to rely on the cut-and-try method to get the exact value you need" is indeed quite correct. But in addition to the value of beta showing a large variation from sample to sample, it is highly temperature-sensitive.

If you are really concerned about stability, you will also need to ensure that the power transistor is held at a fairly constant temperature. The current gain of a common-emitter stage such as this, with no local feedback, varies by about 0.5% per degree Kelvin.

There is also another source of instability that can be significant at high temperatures; namely, the collector-base leakage current. Any effects due to this latter current are exacerbated by the current-source feed to the base, with no other base-emitter resistance present. This current is the same polarity as the current-source current and has a temperature dependence that is exponential, doubling for about every ten degrees Kelvin rise in die temperature. However, unless the heatsink is very small or the ambient high, this term will likely be small compared to filament-size currents.

A far better approach to current regulation is to use a small-value resistor placed somewhere in series with the filament load, and a feedback loop contrived to compare the voltage drop thus developed to a suitable low voltage reference. The effective 1.25V internal reference of the LM317 could serve as part of a power-transistor-assisted circuit, as well as other low-power voltage references such as the TL431 (with perhaps its 2.495V reference suitably divided down).

Depending on where you place the resistor, you could control the error in the filament current wholly by the accuracy of the voltage reference and resistor tolerance(s), or at worst by the alpha (common-base current gain) of the power transistor, which varies with the much smaller tempco of about 0.5% per degree Kelvin divided by the beta of the device. Use a power MOSFET and there will be no control current error wherever the resistor is located.

If anyone is interested in sample schematics they can send me an e-mail and I'll try to grind one out.

Since, for constant emissivity, the goal is to run the filament or cathode at a constant temperature, another intriguing possibility is a power regulator, whose feedback is based on computing the product of voltage and current. It could also incorporate compensation for ambient temperature as well. However, this is an appreciably more complicated design problem.

Brad Wood
swamipoin@yahoo.com

*Michael Kornacker responds:*

*Thank you, Mr. Wood, for your response to my article in which I showed how to use an LM-317 current regulator circuit driving a high power pass transistor in order to regulate current loads greater than what the LM-317 could regulate by itself.*

*I must agree with you because bipolar power transistors such as the 2N3055 do have thermal stability issues you must be aware of in designing such circuits as those I offered. Bipolar transistors are notoriously prone to the snowballing effect of thermal runaway if too much current is drawn through them and they're not well heatsinked. Overheating can destroy them.*

*My intent with this circuit was to show an absolutely basic idea to exhibit the concept. However, circuits built for practical applications may need extra design work to counter the effects of heat and ensure stable operation.*

*You say you might be able to improve it? Feel free to do so. I'm sure our kind editor would be interested in publishing such a circuit. And I'll bet there's someone out there in readerland who needs it.*

## WIRE EFFECTS

I don't know whether to laugh or cry when some respected engineer says interconnect wiring doesn't make any difference. I believe that this person has either never experimented with different cables or has a serious hearing problem.

When I got into digital, I bought a bunch of cables (transmission lines) to see what would work best between my transport and outboard gear. In a transmission line, cable inductance and capacitance are arranged so that the cable has a characteristic impedance  in this case 75W . With the correct source and terminating resistors, any properly designed transmission line will do a good job of passing square waves. Except for one, all these cables were one meter long  not long enough for series resistance to have much effect.

So, with all these cables having a controlled LC characteristic and negligible difference in series resistance, a "pure" engineer will tell you they should sound similar, but they don't. You would need to be almost deaf not to hear the difference between an Illuminations D-60 and a cheap surplus model. (And no, I don't own stock in Illuminations or Kimber.)

So, accounting for LC and R in this case still doesn't explain everything. No doubt some scope jockey will expound propagation velocity, but I have a different view of things.

One big piece of the puzzle is entirely non-electrical. Simply stated, alternating current will dissipate mechanically, and there is an optimum amount of mechanical dumping that best serves music.

I'm entitled to my opinion and I will proffer some evidence. I'm 51 and have seen only one sentence about this phenomenon  in the Mogami cable data. It states:

"NEGLEX 8mm series cables come both in a black flexible and blue semirigid version. The cables are identical in measured specification, however, the rigid type prevents micro-vibrations and will be slightly clearer at the lowest frequencies."

I have used both of these "identical" cables as guitar cables, and they do sound different even though they are electrically identical. Another significant feature of this Mogami cable is its stranded wire. It has been my experience that for larger gauge solid conductors, the mechanical considerations increase.

In the early '90s I was using a wire called W20P in my guitar amp experiments. This was an unplated 20-gauge high-purity copper wire with a thick polyethylene jacket, sold by a now-defunct company called Precision Audio Supply. Atma-Sphere, the OTL people in Minnesota, liked it and had it made up special for their amps. They sold me some four or five years ago, so it might still be available, but I've gone on to other things.

This wire seemed to work pretty well for me, and it was cheap. At one point I decided to rewire one of my guitars with it  including the connection between the pickup and constant cavity  only the hole was too small to fit two pieces of W20P. Rather than modify the guitar, I pulled off the thick polyethylene jacket and put on some thinner shrink tubing. This gave enough clearance to make the connection without enlarging the hole.

Only now it sounded hard and harsh. It had worked well in my amp, but now sounded much worse in my guitar. As an experiment, I enlarged the hole and rewired with W20P along with its regular outer jacket. With no other changes, it sounded much better.

I still had no explanation for this, but upon reflection all I could think of was there was something mechanical going on between the outer jacket and copper conductor. Further experiments including constraining the outer PE jacket with shrink tubing produced positive results. I suggest two layers of shrink tubing if the W20P is carrying DC and one layer if it's just carrying an AC signal.

There must be someone else out there who has run into this phenomenon  let's hear from you. It's my belief that this phenomenon is measurable (since it's audible). However, it might be like silver plating on a copper wire: it measures a fraction of a decibel at one frequency, i.e., the measurement, for some reason, doesn't correlate with the subjective difference.

Rick Bergman
Missoula, Mont.

## CORRECTION

While looking over my article on modifying the Dynaco SCA-35 (Aug. '04, p. 46), I noticed an error in Fig. 5. The 47k resistors shown should be 43k instead. Both the text and the parts list have the correct values. I regret that I did not catch the error at the proofing stage.

James Lin
Albuquerque, N. Mex.

## TWEETER FIX

One of my two Janzen tweeters stopped giving sound, so I brought it in to a local repair shop, where they replaced a pot and a transformer. But I still can't get any sound out of it. This is a very old unit, but has been handled with care. Could the problem be in the plates themselves? Are there any replacement parts available? How should I go about debugging this problem? Any help would be appreciated. This is a model 1-30M Janzen.

Sherwin Dubren
sherwindu@comcast.net

*Edward T. Dell responds:*

*Make sure the panels are clear of dust, which tends to accumulate. Also, the power supply polarizes dust out of the air. Check and clean all connections and terminals. The high voltage diode, a TV type, is often a problem. Be sure the series resistors are intact as well. Unless the panels have been punctured they should be functional, if clean. Check the Internet for vendors who specialize in electrostatic speakers.* ❖

# Showcase

## A 70W McIntosh Amp

I wanted to share with you some pictures of a project that you featured in *Glass Audio* 1/90 ("Build 70 Watts of McIntosh Tube Power," by Bruce Rozenblit). I started the project over a year ago when I happened across the output transformers from a 230. The amps I built use a toroid power supply with a large military surplus filament transformer. The chassis are Hammond steel units that have been powder-coated in an antique silver finish.

These are the best-sounding amps I have ever built, and I am very proud of them.

Bruce Brown
Valley Springs, S. Dak.

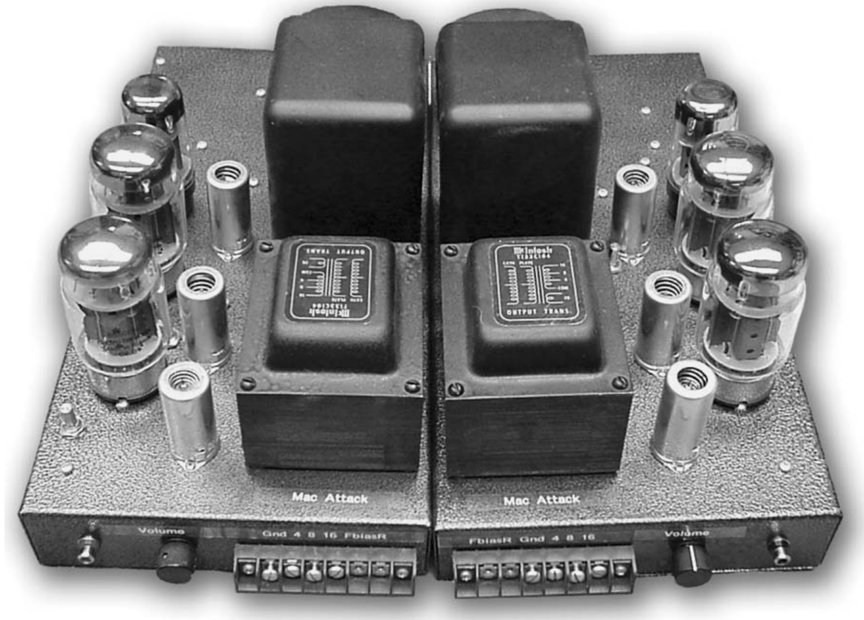



**PHOTO 1: The author's 70W amps.**


**PHOTO 2: A look inside the amp.**

# Classifieds

## VENDORS

## AD INDEX



# Yard Sale

## FOR SALE

NHT 1259 12†woofers, V.G.C. 4/$260 if we meet in New England. 4/$300, plus shipping, if they have to be packed up to ship elsewhere. Call John, 508-835-4490 (weekends only).

Four Eton midrange speakers (4-203/24 Hex), used in excellent working condition. Call for details: 973-579-3372.

Six Gamut D-200 power transformers. Picture available. $45 each, plus shipping. E-mail Ashby, ashby123@comcast.net.

Output transformers, pair, MA230. See Rozenblit's 70W/channel amps in *GA* 1/90. New, in box. Current new prices $149 each. BO over $198 + shipping (18 lbs). Constant current power supply, Electronics Measurement, Inc., model C632. Metered to 500V, 500mA. $50 or BO. Call 03-605-2159 or e-mail: rorseer@yahoo.com.

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

# Audio-Optical Isolation Amp

This circuit enables the full galvanic isolation of sound-frequency signals with optical transmission in excellent quality, eliminating interference. **By Jenoe Keceli**

When you connect audio components, interference may occur from various power and signal grounds. For safety reasons, screenings are interconnected through the protection ground and the protection wire. The circuit that is formed this way opens a free path to the noise and hum from the power supply network and to the interference that originates from computers and other high-frequency generating equipment.

By eliminating protective grounds, you could reduce the problem, but this is not allowed, because all power equipment must have a protection ground. Disabling the already existing protection ground can be hazardous. But it is not necessary to sacrifice safety to construct a system free of ground loops and annoying hum. By separating the reference points, you eliminate the potential differences and disturbing current loops, thus hindering hum and noise.

For many decades the solution for eliminating interference was the isolating transformer; however, this solution was rather problematic. You can only realize the complete audio frequency response with an extra compensated amplifier. This frequency response can be worsened by possible resonance responses; furthermore, the compensation amplifier is sensitive to hum from the power supply network. Correct impedance matching requires additional amplifier stages, resulting in expensive equipment.

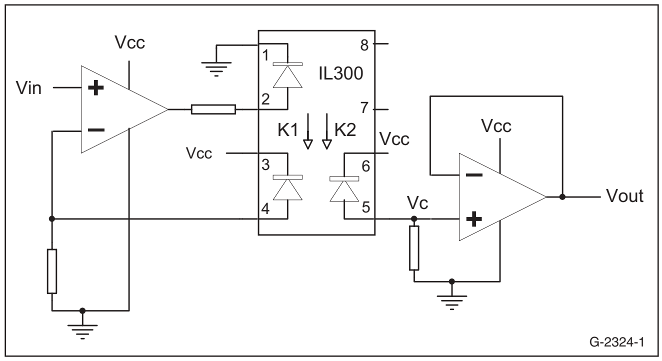



**FIGURE 1: A typical application circuit of IL300.**

It was only with the development of optoelectronics that a new solution has emerged. With an ordinary optocoupler, you can transmit only digital signals. However, the characteristics of the optocouplers have been improved so that they enable linear transmission of the applied signals.

## THE APPLIED OPTOCOUPLER

In this audio frequency application, I have chosen the IL300 linear optocoupler, which consists of an IRLED that is irradiating an isolated feedback and

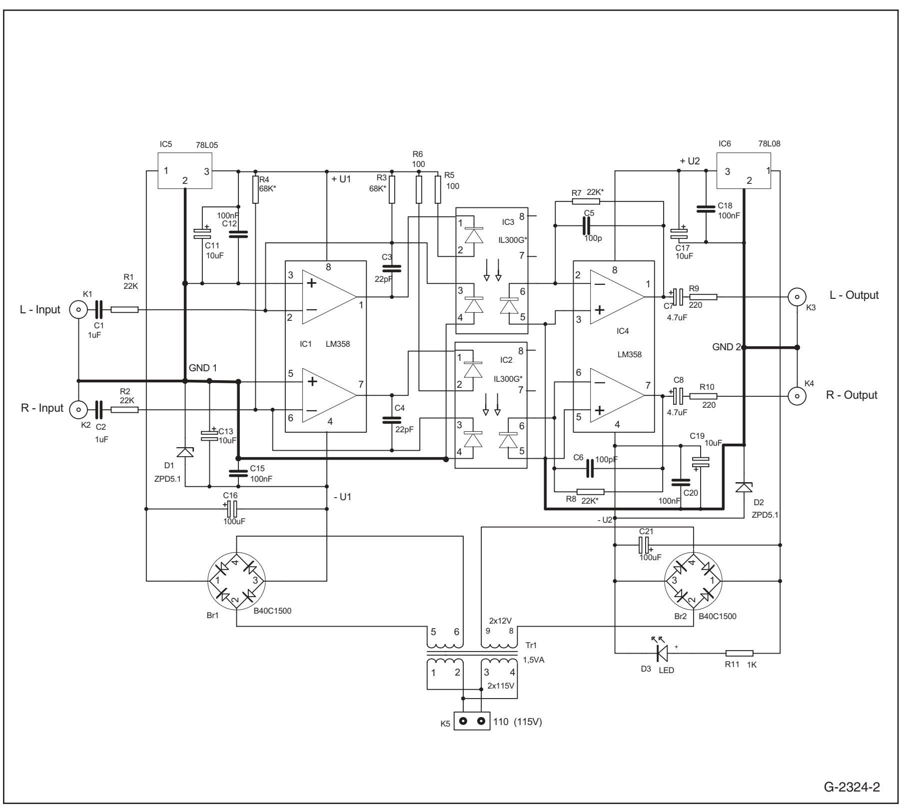



**FIGURE 2: This solution is appropriate for applications external to the equipment because it is supplied from 110V AC.**

### ABOUT THE AUTHOR

Jenoe Keceli is a long-time electronics enthusiast with 40 years of experience. His main fields of interest are designing and building tube and transistor power amplifiers. He has much experience with sound and disco equipment. His articles have been published in electro technical journals. He now runs his own firm, where he builds, repairs, and gives advice (e-mail: kecelijeno@ freemail.hu).

an output PIN photodiode in a bifurcated arrangement. The feedback photodiode captures a percentage of the LED's flux and generates a control signal that can be used to servo the LED drive current. This technique compensates for the LED's nonlinear characteristics. The output photodiode produces an output signal that is linearly related to the servo optical flux produced by the LED.

A typical application circuit (*Fig. 1*) uses an operational amplifier at the circuit input to drive the LED. The feedback photodiode sources current to R1 connected to the inverting input. The output photodiode is connected to a noninverting voltage follower amplifier. The photodiode load resistor, R2, performs the current-to-voltage conversion.

The photodiode is operating as a current source. The output current is proportional to the optical flux supplied by the LED emitter. The receiver diodes can be applied either as a photovoltaic or as a photoconductive source. In the case of the photovoltaic configuration, the linearity is better, there is less noise, and, as a result, the configuration is more stable. However, in the case of photoconductive mode, frequency response is better.

As in the case of sound frequency applications, there are no special requirements for the frequency response, so my choice was the photovoltaic solution. With IRLED, the best linearity can be obtained at drive currents between 5mA to 20mA. The LED is functioning in an optically controlled circuit, thus linearizing the light emission and minimizing the effects of aging and temperature changes.

The photocurrent is proportional with the light intensity of the inner IR diode and feedback diodes. The IL300 is available in various groups according to light-transmission characteristics. This is marked with K3 coupling factor, which is actually the ratio of the coupling factors of the two photodiodes (K3 = K2/K1).

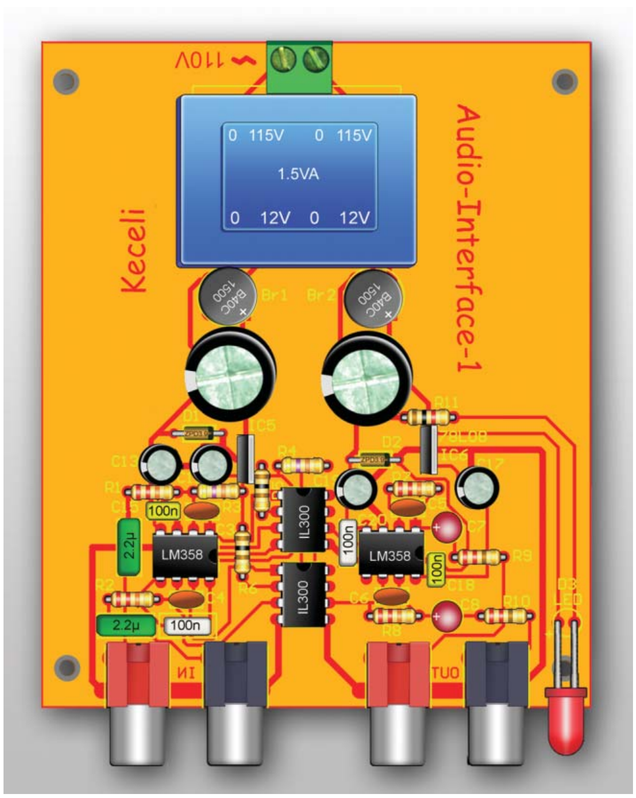



PHOTO 1: The audio interface (transformer-fitted circuit).

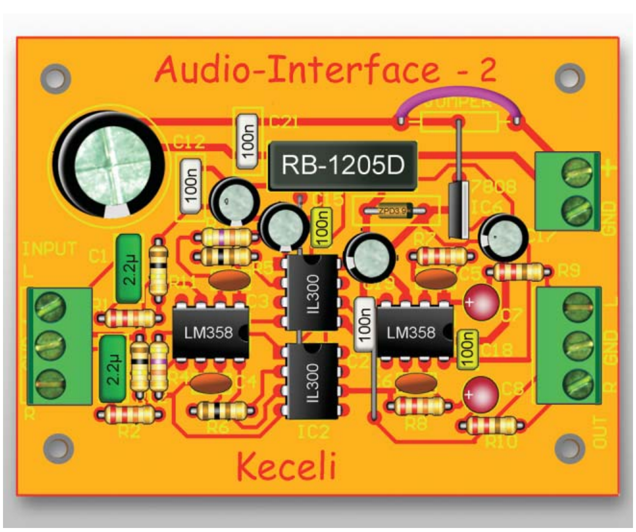



PHOTO 2: The audio interface (for retrofitting).

*Table 1* shows the coupling factor of IL 300s, sorted into K3 bin, indicated by an alpha character that is marked on the part. The bins range from A through J. These optocouplers provide either an inverting or non-inverting transfer function, based upon the type of input and output amplifiers. The non-inverting input amplifier requires the use of a bipolar supply, while the inverting input stage can be implemented with single supply operational amplifiers that permit operation close to ground.

In order to achieve thorough isolation, the power supplies must be separated. Two kinds of applications have been developed. The first solution is standalone equipment with its own 110V AC power supply. The second solution is the installation of the galvanic isolator stage into the system, (e.g., an amplifier), and accordingly can be supplied with DC from the equipment itself.

**TABLE 1**

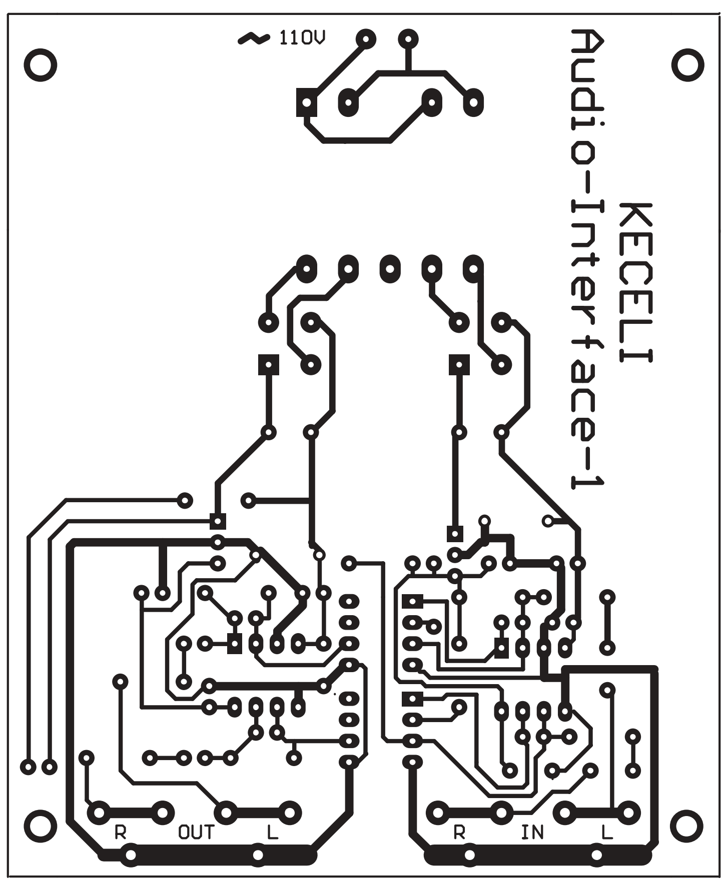
**COUPLING FACTOR OF IL 300S K3 (TRANSFER GAIN) IS SORTED INTO BINS, AS FOLLOWS:**

| BIN | K3 |
|---|---|
| A | 0.557 – 0.626 |
| B | 0.620 – 0.696 |
| C | 0.690 – 0.773 |
| D | 0.765 – 0.859 |
| E | 0.851 – 0.955 |
| F | 0.945 – 1.061 |
| G | 1.051 – 1.181 |
| H | 1.169 – 1.311 |
| I | 1.297 – 1.456 |
| J | 1.442 – 1618 |

K3 = K2/K1 is tested at IF = 10mA

## TEST RESULTS

| | |
|---|---|
| Ordinary audio levels | 775mV eff |
| Maximum audio levels (in and out) | 1V eff |
| Connectors | Stereo input and output<br>Commercial phono—socket (transformer-fitted circuit)<br>PCB mounting terminal block (for retrofitting) |
| Gain | 0dB |




G-2324-3

**FIGURE 3: The PCB is a single-sided board; soldering side (transformer-fitted circuit).**

| | |
|---|---|
| Frequency response | 20Hz...30kHz, practically linear |
| Distortion | <0.05% @ 1kHz, 775mV eff |
| Power on indicator | LED (first solution) |
| Supply voltage | 110V AC (first solution) 14...28V DC (second solution) |
| Current drain | 35mA (second solution) |
| Circuit boards | 104 × 86mm PCB-single sided (transformer-fitted circuit) 80 × 60mm PCB-single sided (for retrofitting) |

**Warning!** Because the equipment operates at 230V, please heed the descriptions and instructions referred to in this section for life-safety reasons.

## PRINCIPLES OF OPERATION—FIRST SOLUTION

This solution is appropriate for applications external to the equipment because it is supplied from 110V AC (*Fig. 2*). A transformer with two separate secondary coils can be found in the power supply so the thorough isolation is assured. The two power-supply sections have similar construction (the only difference is their regulators).

After rectification and filtering, voltage regulations can be found for the input stage 78L05 and for the output stage 78L08. As both sides need an approximately symmetric supply voltage, this is achieved by zener diodes, implementing them in the ground leads of the regulators just mentioned.

As the power

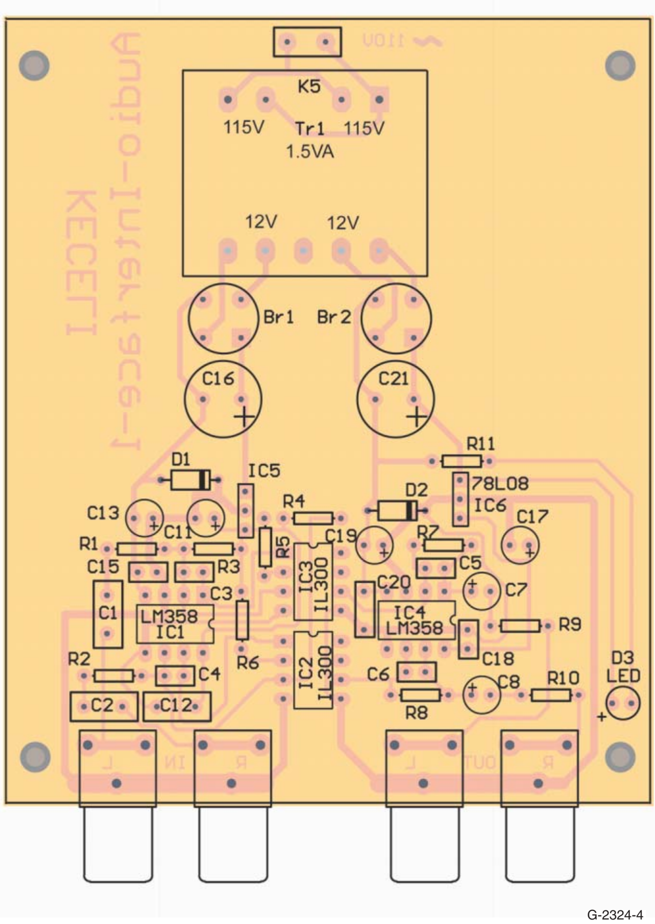



FIGURE 4: The placement of the components; components side (transformer-fitted circuit).

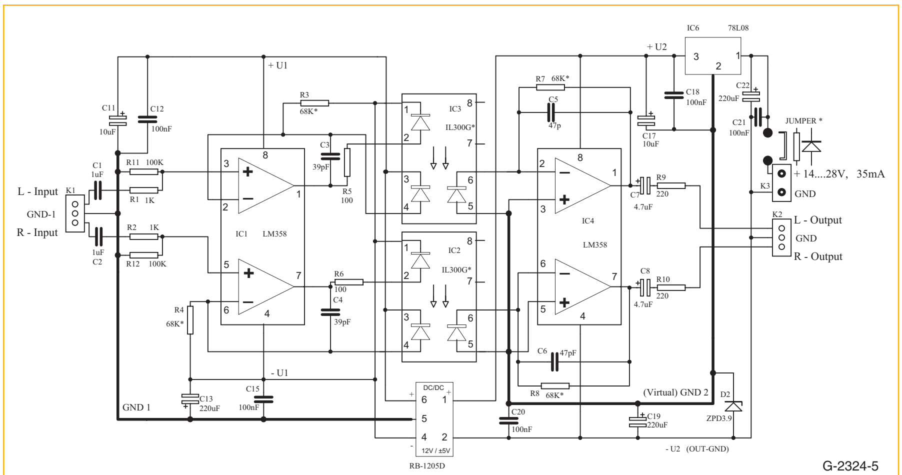



FIGURE 5: The second galvanic isolation amplifier is planned for installation, e.g. in an amplifier or other equipment.

consumption is small, this can be easily implemented for production of the required −5V. Components C13, C15, C19, and C20 filter the noise from the zener diodes. At the regulator output, C11, C12, C17, and C18 filter the remains of probable hum and eliminate high-frequency components, thus minimizing the probable noise of the voltage regulator. The preamplifier gets ±5V supply voltage, while the output amplifier is supplied with +8V and −5V. D3 LED, which is supplied through R11, functions as a power-on indicator.

For the transmission of bipolar audio signals, the approximate bias of both OPs must be ensured. This can be achieved by the R3 (R4) resistors at the preamplifier and the photodiodes that function in photovoltaic mode in the optocoupler. The value of R3/R4 depends on the optocoupler's K3 transmission factor, thus determining the symmetric signal transmission and the most favorable operating point of the IR diode, which in this case is approximately 10mA.

In the case of IL 300G, K3 = 1.1 (approximately); accordingly, the resistor value R3 (R4) = 68k. The voltage of the photodiode is connected to the inverse input (IC1) as feedback, thereby stabilizing the operating point and supporting the distortion-free signal transmission. IRLED current is limited by R5 (R6) resistors. The preamplifier is functioning in inverse mode, but as the signal is connected to the inverse input, in-phase signal appears on the IRLED.

On the receiver side, the voltage generated on the photodiode is amplified by IC4. Balancing the operating point is required here as well, which is achieved with R7

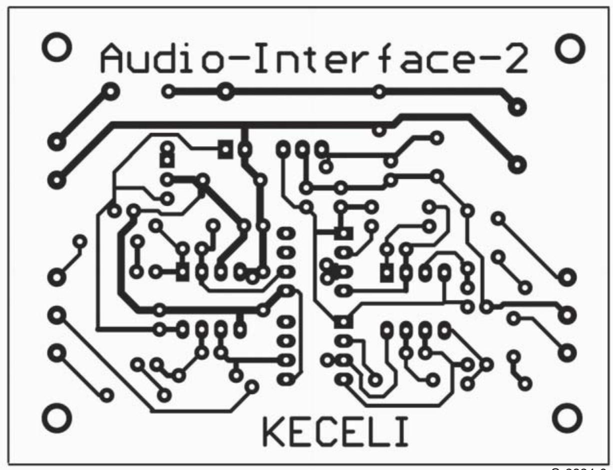



**FIGURE 6: The PCB for the second solution is a single-sided board; soldering side (for retrofitting).**

**TABLE 2**
**PARTS LIST OF THE AUDIO INTERFACE, FIRST SOLUTION**

| SCHEMATIC REFERENCE DESIGNATOR | PART TYPE VALUE | DESCRIPTION |
|---|---|---|
| IC1,IC4 | LM358 | |
| IC2,IC3 | IL300G* | |
| IC5 | 78L05 | |
| IC6 | 78L08 | |
| Br1,Br2 | B40C1500 | |
| D1,D2 | ZPD5.1 | |
| D3 | LED | |
| Tr1 | 2×12V 1.5VA Transformer | Dagnall el.D2014, Farnell Order Code 899–744 |
| K1,K3 | Commercial Phono-sockets | PCB-mounted, red |
| K2,K4 | Commercial Phono-sockets | PCB-mounted, black |
| K5 | terminal block | PCB mounted 2 pole |
| R1,R2 | 22k | ¼W |
| R3,R4 | 68k* | ¼W |
| R5,R6 | 100Ω | ¼W |
| R7,R8 | 22k* | ¼W |
| R9,R10 | 220Ω | ¼W |
| R11 | 1k | ¼W |
| C1,C2 | 1µF/16V | MKS, MKP |
| C3,C4 | 22pF/40V | Ceramic |
| C5,C6 | 100p/40V | Ceramic |
| C7,C8 | 4.7µF/16V | MKS, MKP, Tantalum or Electrolytic-Radial |
| C11,C13,C17,C19 | 10µF/16V | Electrolytic-Radial |
| C12,C15,C18,C20 | 100nF/40V | Ceramic |
| C16,C21 | 100µF/16V | Electrolytic-Radial |
| PCB | 104 × 86mm | Single Sided |

* refer to the text

(R8). These resistors set the feedback, determining thus the level that is identical at the input and the output. The full gain is 0dB.

If required, you can set the amplification by changing R7 (R8). Components R9 (R10), which are found on the output, protect against overload and probable high-frequency noise. The amplifier is compensated (by neutralization) against noise with C3, C5 (C4, C6) capacitors. Because the equipment is designed for independent application, commercial phono sockets are fitted at the input and the output.

## CIRCUIT ASSEMBLY—FIRST SOLUTION

The PCB is a single-sided board as shown in *Fig. 3. Figure 4* shows the placement of the components. The board is slightly longer, so the electrical network transformer can fit farther away from the amplifier components in order to avoid hum. All the components, the transformer, and the connectors are assembled on the PCB. There are no jumpers on the PCB.

After assembly the equipment should function without any problem. By using the IL300-G optocoupler, there is no need for further adjustments after the assembly. Unless you use an optocoupler with specific transfer gain G, you must select the values of the resistors R3 (R4) on the basis of the coupling factor K3. The correct value for IL300 can be determined by calculation (refer to the IL300 product specifications), but it is better to ascertain it by measurement. For this test you need an oscilloscope and a tone generator.

An approximately 1V eff, 1kHz signal is applied to the input of the galvanic isolation amplifier. At the same time, the balance of the output signal is monitored by the oscilloscope. A

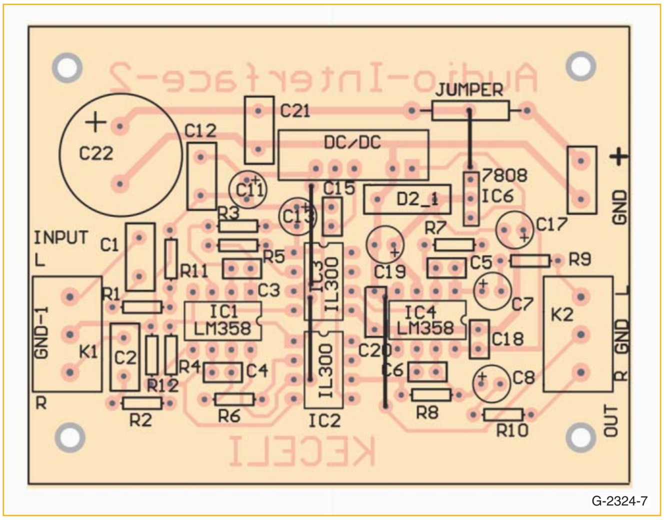



FIGURE 7: The placement of the components, components side (for retrofitting).

### TABLE 3
### PARTS LIST OF THE AUDIO INTERFACE, SECOND SOLUTION

| SCHEMATIC REFERENCE DESIGNATOR | PART TYPE VALUE | DESCRIPTION |
|---|---|---|
| IC1,IC4 | LM358 | |
| IC2,IC3 | IL300G* | |
| IC6 | 78L08 | |
| Br1,Br2 | B40C1500 | |
| D2 | ZPD3.9 | |
| DC/DC | RB-1205D | 12V/±5V 1W DC/DC Converter |
| K1,K2 | terminal block | PCB mounted 3 pole |
| K3 | terminal block | PCB mounted 2 pole |
| R1,R2 | 1k | ¼W |
| R3,R4 | 68k* | ¼W |
| R5,R6 | 100Ω | ¼W |
| R7,R8 | 68k* | ¼W |
| R12,R11 | 100k | ¼W |
| R9,R10 | 220Ω | ¼W |
| Jumper* | Jumper, 1N4148, or resistor | |
| C1,C2 | 1µF/16V | MKS, MKP |
| C3,C4 | 39pF | Ceramic |
| C5,C6 | 47p | Ceramic |
| C7,C8 | 4.7µF/16V | MKS, MKP, Tantalum, or Electrolytic-Radial |
| C11,C17 | 10µF/16V | Electrolytic-Radial |
| C12,C15,C18,C20,C21 | 100nF | Ceramic |
| C13,C19 | 220µF/16V | Electrolytic-Radial |
| C22 | 220µF/35V | Electrolytic-Radial |
| PCB | 60 × 80 mm | Single Sided |

* refer to the text

perfect, symmetric, sine-wave signal, without clipping, must be adjusted on the output. This is carried out by temporarily changing the values of R3 (R4).

In this case, you can replace these resistors with a serial combination of a 50Klin (Trim) potentiometer and a 22K resistor. (It is sufficient to carry out the measurements on one of the channels, thereby determining the value of the resistor. You can fit the new resistors by soldering in both channels.) If required, you can modify the gain by changing the values of R7 (R8).

Finally, you must mount the galvanic isolation amplifier in a suitable case, which can be either plastic or metallic. However, since the case is connected to the output ground, metallic is a more appropriate style because of the screening.

## PRINCIPLES OF OPERATION—SECOND SOLUTION

The second galvanic isolation amplifier (*Fig. 5*) is planned for installation, e.g., in an amplifier or other equipment. It can be powered from a single-ended, approximately 14–28V DC power supply, which is usually available from the power supply of the host equipment.

There is a significant difference in the supply circuitry of the two solutions. The output amplifier and the voltage stabilizer are similar as in the first solution. The difference is in the application of the symmetrical power-supply stage. The middle point of the supply circuitry is used as the virtual ground, while the output ground is the negative side of the power-supply circuitry. When installing, it is actually not necessary to connect both earthing conduits.

The power supply of the preamplifier is generated by a DC/DC converter. This way a stabilized ±5V is produced from 12V, which is regulated by a 78L08 IC increased with the voltage of the D2 zener diode. Because a high-frequency DC/DC converter is used, more effective filtering is needed than in the first solution. This is why the capacity of the C13 and C19 capacitors is 220µF.

The input amplifier operates in non-inverting mode, which is different than in the previous solution. Hence the amplification of the input stage is lower. This should be compensated for on the output by increasing the value of R7 (R8) to 68k to achieve the target 0dB gain. The IC2 connection of the output stage is the same as in the case of the first solution. It is an inverting amplifier but with photodiodes in photovoltaic mode, hence there is no phase inverting.

## CIRCUIT ASSEMBLY—SECOND SOLUTION

The PCB for the second solution is shown in *Fig. 6*. *Figure 7* shows the installation of the components.

Because there are four jumpers on this board, you must first solder these in. The order is important because some jumpers are placed under the ICs. When finished, you can solder in the other passive components and finally the semiconductors.

There is a component marked "jumper" on the PCB in the power supply stage. If the value of the power supply is less than 28V, you can jumper this component or solder in a diode for polarity protection. In case of higher supply voltage, you can use a serial resistor. The current consumption of the isolation amplifier is approximately 35mA, and the serial resistor must be designed accordingly. The applied DC/DC converter does not cause any interference.

After assembling, the isolation amplifier should function immediately. If an optocoupler with different transfer gain is used instead of the one indicated on the drawing, the values of R3 (R4) must be as outlined in the first solution. If needed, you can change the values of R7 (R8) to achieve higher gain. Because this module is planned for retrofitting, I applied screw terminal blocks as connectors. ❖

## REFERENCES


**Infineon**, IL300 Linear Optocoupler—applications, www.infineon.com/opto.

**Texas Instruments** production data information LM358.

*ELV Journal* 1999/3 (German)—Optischer Trennverstärker für analoge Audiosignale.

*Elektor Journal* 2003/3 (German)—Übertrager gegen Brummprobleme.

(English)—SAA3049 Replacement, pp 42–48.