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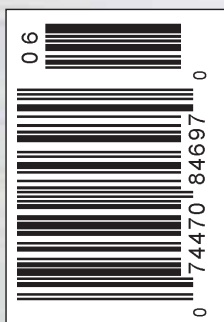
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A Three-Way Tri-Amp Center-Channel System

This article describes the design and construction of a three-way center channel with superior performance.

PHOTO 1: Center-channel system.

I had been considering a center-channel speaker project for some time as a potential upgrade to my theater system. My existing unit was one of those rather generic M-T-M configurations with 4" midwoofer drivers. The tonal characteristics of the center seemed quite different from that of the main left-right speakers that are custom made and driven with an active crossover through three separate amplifiers. But was it worth the extra effort and expense of a revamped crossover design and additional amplifier channels to construct?

I'm happy to report that I found much improved performance by constructing a three-way center speaker design. This article describes one possible approach to that configuration.

RESEARCH

The first step was to measure the existing center-channel speaker and compare on-axis readings with 30° off-axis measurements (Fig. 1). (The Appendix explains the measurement method.) Two problems become immediately apparent. The first problem is the pronounced lift in the high-frequency range, which equates to an overly bright sound when compared to the main speakers. Some designers apply a high-frequency lift to balance a high-Q bass alignment¹.

The second problem is that the severe off-axis drop in response in the 1400Hz region reduces intelligibility. There were instances in the past when listeners (sitting off-center from the display) requested an

increase in volume to an already reasonably loud movie soundtrack because of trouble understanding the dialogue. Although the main speakers are placed symmetrically in

the system, the display is positioned somewhat off-center due to a difficult-to-overcome WAF (Wife Acceptance Factor).

I continued by establishing a simple

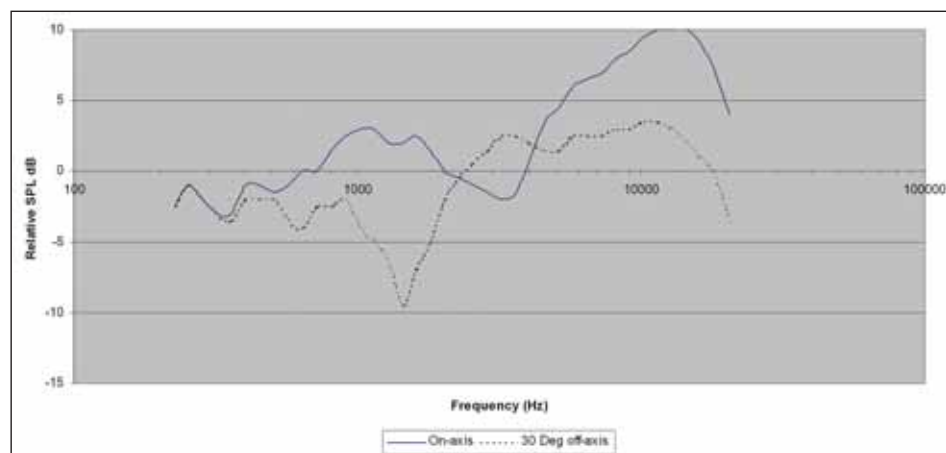


FIGURE 1: Existing center-channel response.

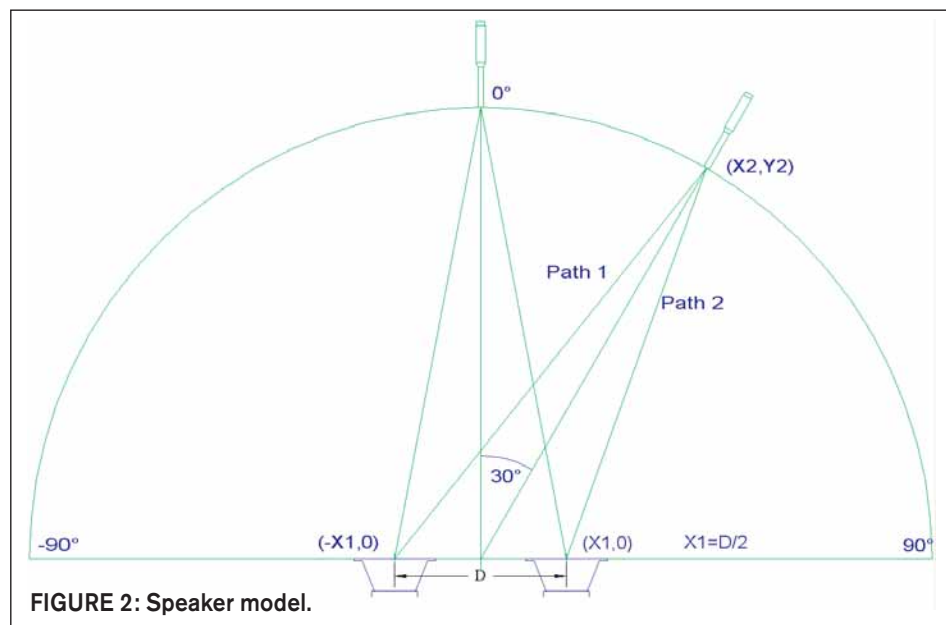


FIGURE 2: Speaker model.

model of the existing speaker to better understand the anomaly or possible design compromise existing in this unit (Fig. 2). Two drivers are radiating into the horizontal plane and separated by distance D. Assume that each driver is emitting an identical sine waveform that individually would produce a 90dB/1m reading on an SPL meter (on-axis). A measurement microphone is kept at a constant distance of 1m and may be moved from 0° to 90° in the horizontal axis. At the 0° mark the difference in

path lengths to each driver is zero. The identical waveforms will add together and produce a level of 96dB.

As you move the microphone from the 0° to the 90° marks, the path length difference increases to a maximum of D and results in the signal from one driver arriving at a different time to the microphone. This time delay is expressed as a phase shift relative to the wavelength of the signal and results in an addition or even cancellation of sound. The method for predicting the response is as follows:

To calculate the path lengths from the drivers, you first need to calculate the x and y values for a 1m (39.4") circle. As the x-axis on the graph proceeds from -90° to +90° (left to right), the equivalent polar angle goes from 180° to 0°.

$$x = r \times \cos \theta$$

$$y = r \times \sin \theta$$

where $r = 39.4''$.

The path distance to each driver is found by using the equation for a length of a line:

$$\text{path}_{1,2} = \sqrt{(x_2 - x_1)^2 + (y_2 - y_1)^2}$$

Use the path length difference to find the phase delay in seconds:

$$\theta_D = \text{delay(s)} = \Delta \text{path} \left(\frac{1s}{13560in} \right)$$

The phase delay in radians is expressed as²:

$$\theta_{RAD} = \theta_D \times 2\pi f$$

Finally, the combined levels of the two signals are found by² (angles are in degrees):

$$L_P = 20 \log \sqrt{\left(10^{E1/20}\right)^2 + \left(10^{E2/20}\right)^2 + 2\left(10^{E1/20}\right)\left(10^{E2/20}\right)(\cos(a_1 - a_2))}$$

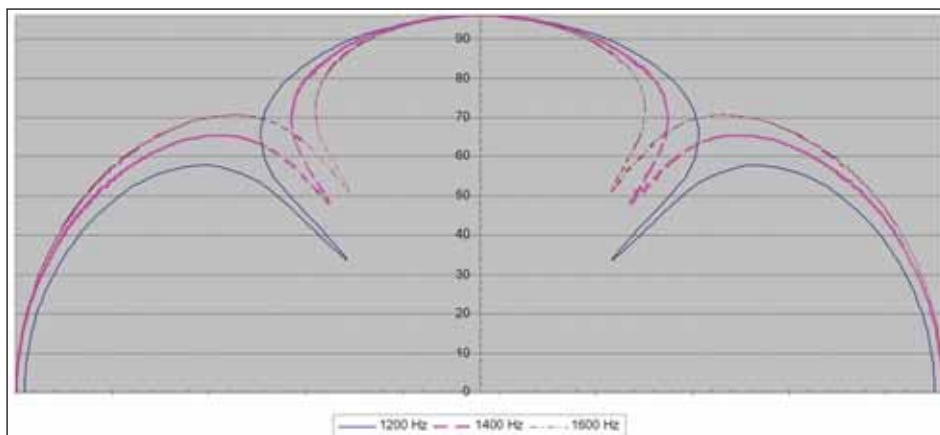


FIGURE 3: Predicted horizontal response (driver separation = 9").

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To complete the response graph, use the same equations to find the (x, y) values of the reference circle above, except that you replace the radius r with the magnitude of the combined levels. Note that the visual proportions on the Excel XY Scatter Plot are not quite the same as they would be on a true polar plot, but are adequate for this exercise.

Figure 3 shows the predicted polar response of the existing M-T-M speaker at the frequencies of 1200, 1400, and 1600Hz. The dip in response at 1400Hz occurs at $\sim 30^\circ$ and correlates to the actual measurements taken. This suggests that the off-axis coloration is due to the interference pattern produced by the two midwoofer drivers. D'Appolito states that odd-order filters can reduce these variations, but recommends that these designs be used in the vertical orientation³. It is doubtful that there was much intent to correct this in the design, because the existing M-T-M had only a simple 1st-order crossover filter network.

DESIGN AND CONSTRUCTION

I was now eager to construct a new center channel to bring performance in line with the main channels. **Figure 4** details the construction of a three-way, tri-amplified center-channel speaker. By including a small 3.5" midwoofer into the mix, you can eliminate a few issues at the expense of complexity. The small diameter of the midwoofer allows you to mount the tweeter directly above, while at the same time minimizing the height. This facilitates the horizontal orientation of the speaker, which is important for mounting the speaker directly above or below the display or screen.

The midwoofer operates in the range where the interference between the two 7" woofers would cause unacceptable coloration of the sound. **Figure 5** shows the modeled polar response of the two 7" drivers on 15" centers operated at 300, 900, and 2700Hz. At 300Hz the signals add together on-axis and gently roll off toward the 90° mark. At 2700Hz the interference pattern is quite complex and will be avoided with the three-way design. It's interesting to note that this is in the vicinity of a typical crossover frequency for a two-way speaker.

The crossover schematic is shown in **Fig. 6**. This crossover is adapted from the design for the main channels using the same circuit pattern. The input buffer stage uses a low-noise audio op amp and has an optional shelving filter (not used for a three-way design). The fourth-order Linkwitz-Riley filters used here are of the unity-gain Sallen-Key type. The LF347B op amps used here have a more than adequate (2.2MHz minimum) gain-bandwidth product to ensure proper filter performance.

I chose a 300Hz cross point for the Peerless midwoofer, which, at 300Hz, can play sufficiently loud (105dB/1m) without excessive distortion. Also, as stated previously, the polar response of the two woofers is both wide and smooth in their operating range (80-300Hz). The 2.3kHz crossover frequency used for the high-mid drivers is a carry-over from the main system, because there was no practical reason to change it. I used a 3rd-order all-pass filter design to provide a group delay of 80μs for frequencies 0<f<6kHz to the tweeter. This maintains the symmetrical vertical polar response pattern between the high-mid drivers.

Note that the delay value is determined by playing a tone burst signal at the crossover frequency⁴. The time difference between signals arriving from the tweeter and midwoofer

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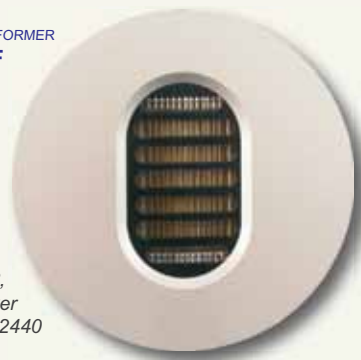
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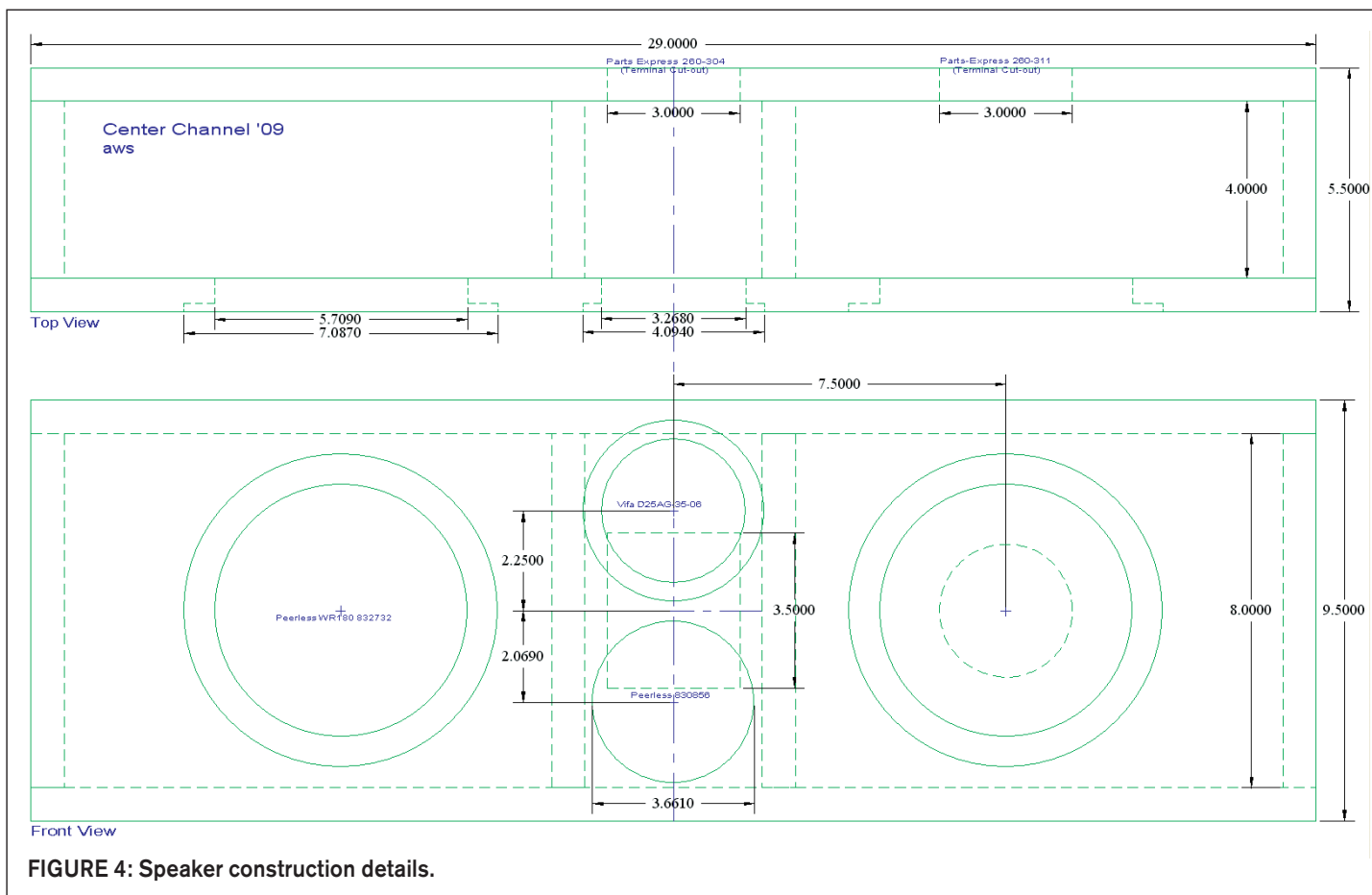


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A feature of the crossover is the use of a shelving filter (C6, R6) in the inverting output stage of the all-pass filter section. I have found it useful to reduce the tweeter output at frequencies well above the cross point (-5dB at 8.4kHz) rather than an overall reduction in gain. This avoids contributing to a drop in amplitude around the crossover point.

Input Buffer/ RF Filter with
Optional Shaking Filter

ER

C2 10uF

R1 2.00k

R3 10.0k

R2 2.00k

C3 0.001uF

U1B OPA2134

U1A OPA2134

C1 0.1uF

C4 0.1uF

R4 Rshk1

R5 Rshk2

C5 Cshk

J2 2-way Crossover Op

J3 JUMPER

J2 JUMPER

Shaking Filter Option

I solidly constructed the speaker from ¾" MDF and flush-mounted all drivers with the use of a plunge router and circle guide. The center chamber for the 3.5" mid is approximately 0.065ft³. The separate chambers for the Peerless 7" woofers are 0.20ft³ each. The target sealed-box alignment for both types of drivers is a Q_{ts} of 0.7. I used fiberglass insulation for 100% fill to absorb internal reflections.



I connected both the tweeter and midwoofer through a bi-amp terminal in the rear center of the speaker, and connected the woofers together in series to maintain the sensitivity equal to that of a single woofer. The speaker wire

connecting the 7" drivers runs through the bottom of the middle chamber from left to right. Apply RTV silicone to the entrance and exit of these wires to keep the chambers sealed.

And finally, **Fig. 7** shows the measured

response of the new center-channel project. The curve is in stark contrast to the one taken of the commercial M-T-M. The following table contains the range and standard deviation of the measurements taken for purposes of comparison:

	Range, dB		Std Dev
	On-Axis	30° Off-Axis	
Existing Center	13	13	4.12
New 3-way Center	6.5	7	1.78
	30° Off-Axis	7	1.86

Response curves with a standard deviation of one are perceived to have the best sound quality¹.

My subjective description of the performance is that the speaker has a more natural presence and consistency of sound than the predecessor. I was very satisfied with the way that this project turned out and am happy to say that the *WAF* is relatively high as well. *aX*

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APPENDIX

The testing performed here is based on the shaped tone burst method described by Linkwitz⁴. This method facilitates the equivalent of anechoic measurements as long as the signal emitted by the speaker arrives at the test microphone before any reflections. That is why the graphs are limited to a frequency just above 200Hz. A microcontroller circuit controls the (raised cosine) envelope and timing of the burst signal. A Behringer measurement microphone receives the signal.

Relative signal levels are determined with the aid of precision 0.5dB step attenuator and an oscilloscope. While this setup may not replace sophisticated test equipment and software, it's inexpensive and adequate for making comparisons and debugging speaker designs. ■



PHOTO 2: Inside the unit.

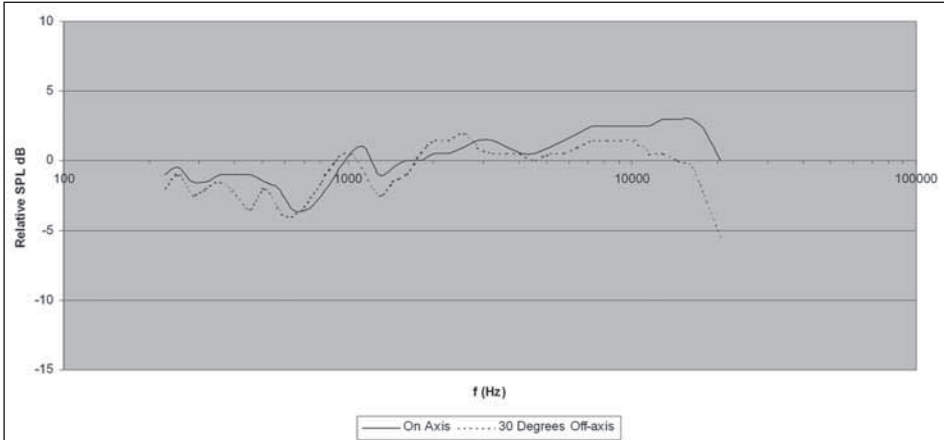


FIGURE 7: Center channel response.

TABLE 1

Crossover Bill of Materials

Reference	Value	Description	Manufacturer	Digi-Key Part #
C1, C4, C11, C12, C19, C20, C28-39	0.1µF	polypropylene	Panasonic	P3104-ND
C2	10µF	electrolytic	Vishay/BC	4211PHCT-ND
C3	0.001µF	polypropylene	Panasonic	P3102-ND
C5	Cshlva	(optional)		
C6	.0015µF	polypropylene	Panasonic	P3152-ND
C7-9, C25	33pF	ceramic	Vishay/BC	1431PH-ND
C10, C13, C18	.0047µF	polypropylene	Panasonic	P3472-ND
C14-17, C21-24, C26, C27	0.01µF	polypropylene	Panasonic	P3103-ND
C41, C40	2.2µF	tantalum	Kemet	399-3550-ND
JP1, JP4, JP5, JP7, JP8	4 header	0.100" centers header	generic	
JP2, JP3, JP6	jumper	NA - wire jumper		
R1, R2	2.00k	1% metal film	Yageo	2.00KXBK-ND
R3, R9, R10, R16, R19, R20, R21	10.0k	1% metal film	Yageo	10.0KXBK-ND
R4	Rshlva1	(optional)		
R5	Rshlva2	(optional)		
R6	12.7k	1% metal film	Yageo	12.7KXBK-ND
R7	3.65k	1% metal film	Yageo	3.65KXBK-ND
R8	4.64k	1% metal film	Yageo	4.64KXBK-ND
R11, R12, R22, R23-25	4.87k	1% metal film	Yageo	4.87KXBK-ND
R13, R26, R27, R32, R37	100	1% metal film	Yageo	100XBK-ND
R14	2.43k	1% metal film	Yageo	2.43KXBK-ND
R15	4.75k	1% metal film	Yageo	4.75KXBK-ND
R17, R18	9.76k	1% metal film	Yageo	9.76KXBK-ND
R28, R29, R33-36	3.74k	1% metal film	Yageo	3.74KXBK-ND
R30, 31	7.50k	1% metal film	Yageo	7.50KXBK-ND
U1	OPA2134	op amp	TI	OPA2134PA-ND
U2-4	LF347B	op amp	National	LF347B-ND

A Loudspeaker That Can Play Square Waves?

This article describes a DSP (Digital Signal Processing) technique developed to correct loudspeakers' frequency domain and time domain distortions.

One of the most useful test signals in electronics is the humble square wave. The "ideal" square wave is a superposition of an infinite number of sine waves, each contributing its required amplitude and phase. It is due to this very feature that, when passed through an audio system, the square wave can reveal time domain performance issues of the system. This is because all of its sine wave components must be passed by the system without time distortion, or different delays, in order to recombine as a square wave at the output of the system under test.

Practically generated square waves have a limited number of contributing sine waves, but the number is still sufficiently large that for audio frequency range testing, you can fully utilize the "almost perfect" square waves.

It is important to realize that the system time delay does not need to be zero. But it must be the same for all frequencies within the audio system passband. Such conditions will be easily met if the system under test has a zero-degree phase response.

REAL-LIFE LOUDSPEAKER EXAMPLE

The system under test here consists of a filter and a loudspeaker in an enclosure. The two components that will introduce time delay are the filter and the combination of driver and the enclosure itself.

To illustrate this, I measured a 12" guitar loudspeaker in a vented box. Its minimum-phase (MP) responses were obtained with the help of an MLS measurement technique (Fig. 1). It is immediately observable that the loudspeaker

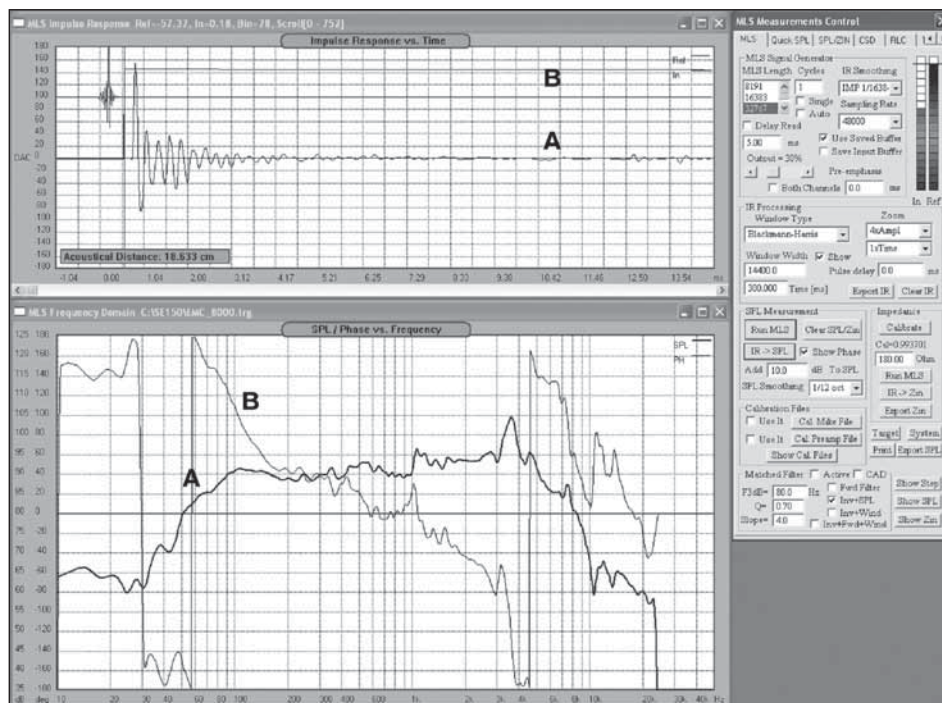


FIGURE 1: Measured system's magnitude (A) and phase (B).

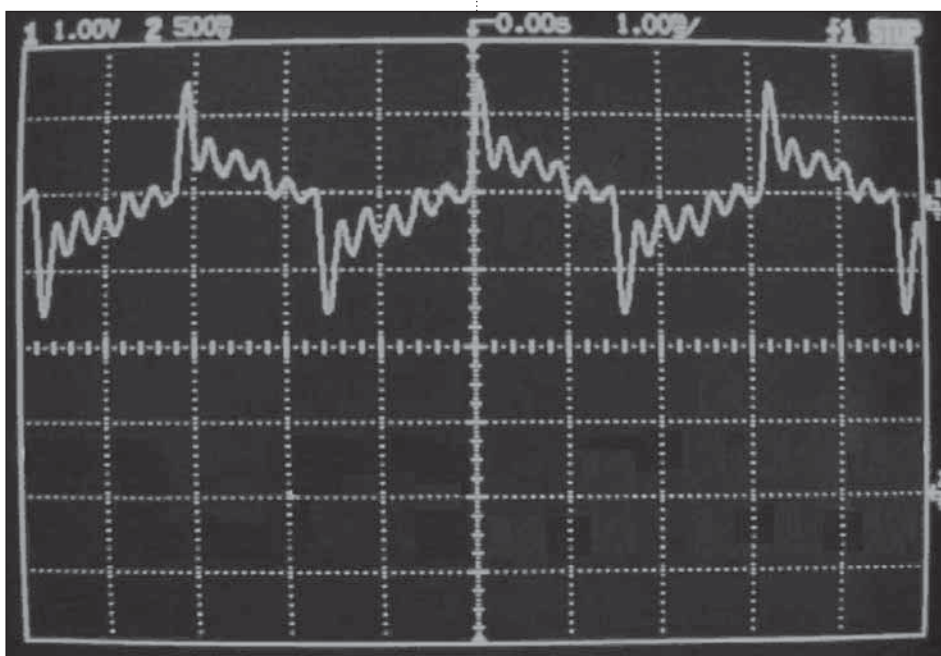


FIGURE 2: Distorted square wave passed through system on Fig. 1.



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has a rather irregular frequency response. Because the loudspeaker is essentially a minimum-phase device, the corresponding phase response is also highly irregular, and definitely not flat.

The *frequency response of interest* is the frequency range where the SPL will be equalized to flat response. In my example, it will be 91Hz–5250Hz. A 300Hz square wave reproduced by this loudspeaker is highly distorted.

The outgoing waveform resembles a triangular-wave with a high level of ringing imposed on it. The ringing is the result of a highly irregular frequency/phase response from 1kHz to 6kHz, with an additional +10dB peak around 3.5kHz. The resulting square wave is shown in **Fig. 2**.

Unfortunately, this is what most loudspeakers do—irregular frequency response, coupled with accumulated delays in the system, prevent the loudspeaker from correctly recombining all sinusoidal components of the square wave. The result is quite poor reproduction of musical transients. Just as the vertical transitions of the square wave and their flat sections are reproduced as spikes

with ringing, musical transitions are also highly distorted.

But musical transients are difficult to capture because they change constantly. More important, you do not really know what the “undistorted” musical transient should look like in time domain (**Fig. 3**).

So, even though nobody listens to square waves, they provide a well-defined test signal that will be very easy to compare with distorted square waves passing through the loudspeaker. The distortions, or deviations, are therefore immediately obvious, and provide clues as to the origins of the distortions. A high level of ringing in the resulting square wave is related to irregular frequency response. This gives the first clue in the quest to improve the shape of the outgoing square wave—flatten the frequency response.

A popular tool used for linearizing a transfer function of an LTI (Linear Time-Invariant) system is a Hilbert-Bode Transform (HBT). Like the Fourier Transform, which allows you to flip between time domain and frequency domains, the HBT allows you to move from magnitude response to phase re-

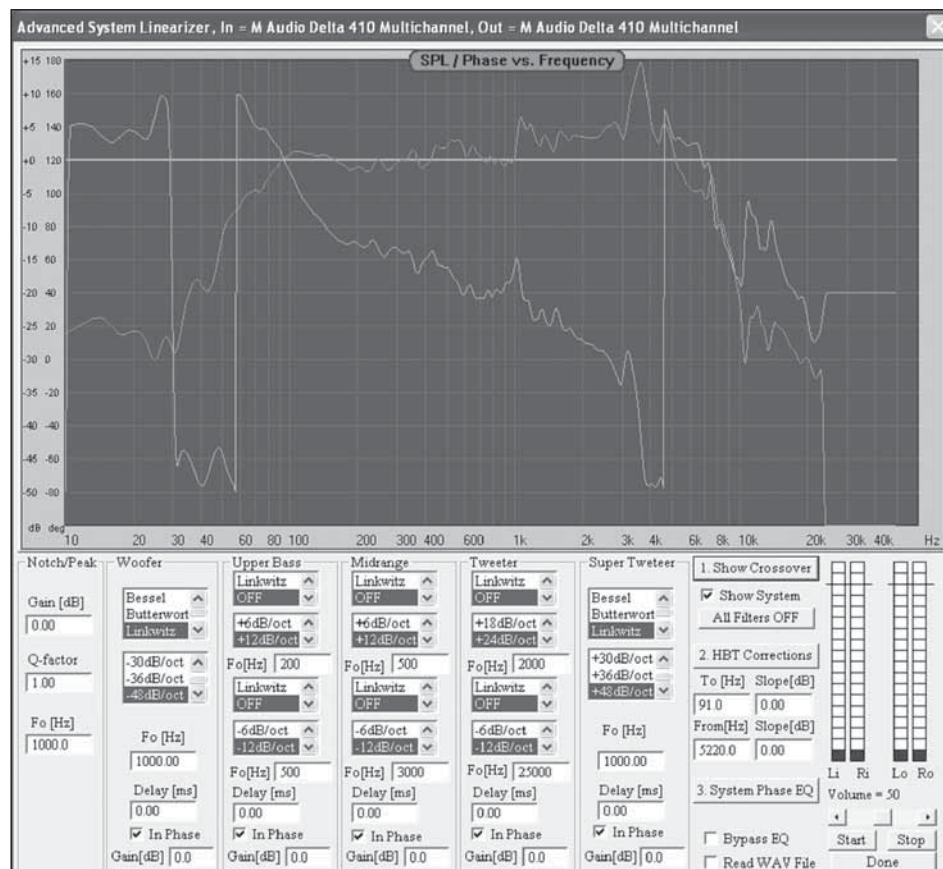


FIGURE 3: Advanced system linearizer controls.

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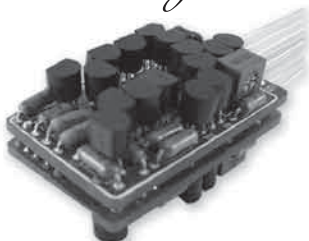
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sponse and vice-versa. I can therefore nominate a frequency range of interest within the loudspeaker's magnitude response, then attach flat "tails" on the low and high side of this frequency range and apply this artificially created magnitude response to the HBT. As a result, I get corresponding phase response, which, in turn, means that I actually have a full complex transfer function calculated via HBT.

ON-AXIS VERSUS OFF-AXIS EQUALIZATION

Typical loudspeakers will exhibit somewhat different frequency response when measured off-axis. **Figure 4** illustrates this situation very well. Because the HBT-based equalizer is being designed for on-axis performance, there is a need to consider its impact on off-axis performance of the loudspeaker.

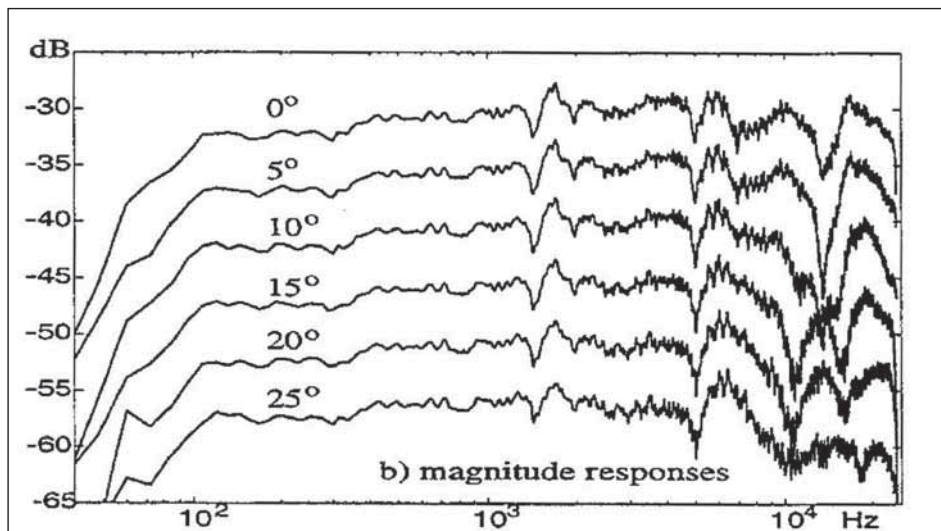


FIGURE 4: Off-axis SPL1.

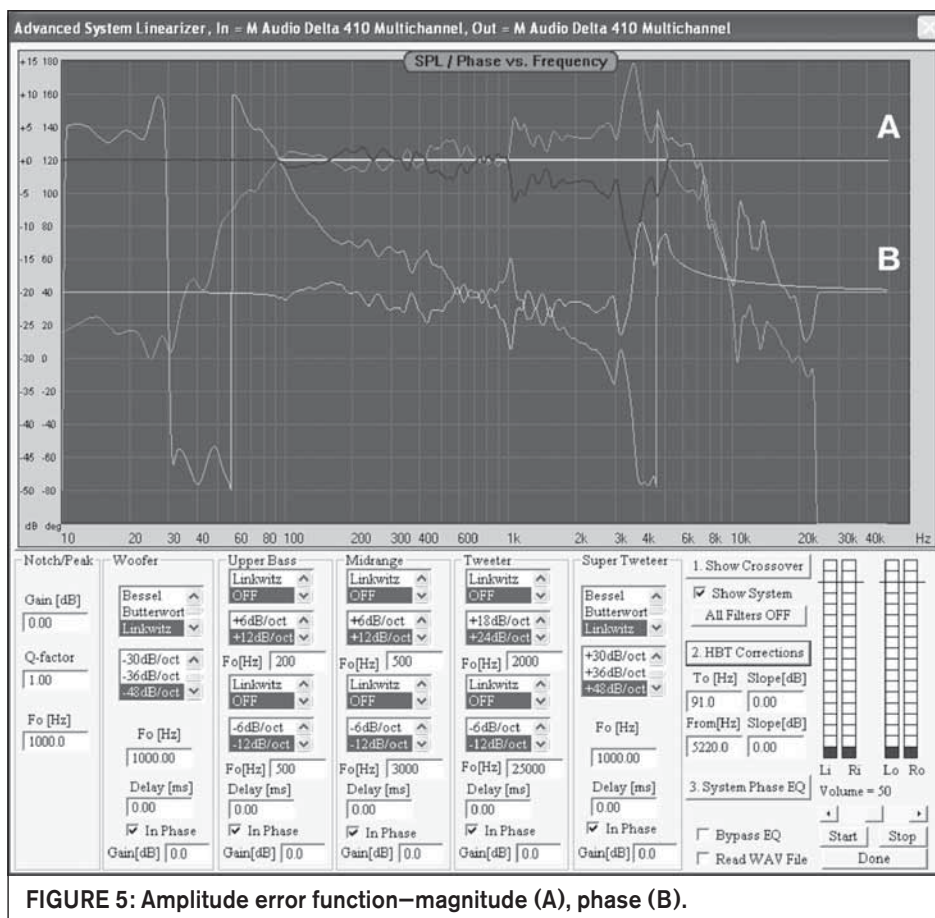


FIGURE 5: Amplitude error function—magnitude (A), phase (B).

It is observable that the 0°, or on-axis, curve is the flattest within the whole family of responses. This is a very common occurrence and is due to the fact that loudspeakers are typically designed and optimized for the on-axis performance. Therefore, the equalizer design for on-axis performance automatically requires the least amount of equalization to perform. This is actually quite often advantageous for off-axis performance as well.

If you examine the off-axis curves on **Fig. 4**, you will notice that as the angle increases, the curves have gradually increased peaks and valleys, but the deviations hold their locations on frequency scale. If the same equalizer were applied to each of those curves, only the on-axis curve would be fully equalized.

Other curves would be equalized to a lesser degree, but they would not be over-equalized. For instance, the 5° curve has a very similar shape as the 0° curve, but the notch at 18 kHz is much deeper. If you correct this curve with the same equalizer, then most of the curve would become almost perfectly flat, but the notch at 18 kHz would still be evident to some degree.

You can examine the curves for other angles and you would most likely conclude that the equalizer would still help correcting many of the irregularities evident in off-axis curves. The above situation is typical; however, not all loudspeakers are as easy to handle as this one. If the peaks and valleys of the off-axis frequency response are unstable in frequency domain, then the effectiveness of equalizer will be reduced.

AMPLITUDE EQUALIZER DESIGN

The importance of having the error transfer function is that it can be convolved with the loudspeaker's transfer function to linearize its frequency response.

The linearization will be accomplished if the HBT-calculated transfer function is first inverted. The result of this operation is a mirror-imaged frequency and phase response calculated within the previously nominated frequency range, and flat response everywhere else.

I have now produced an Amplitude

Error Function. The bold line (A in Fig. 5) is the SPL of the Amplitude Error Function (notice, it's inverted already), and line (B) is the phase of the Amplitude Error Function. Please note mathematically correct phase response and its transitions from irregular-to-flat sections. This is the HBT in action.

Also note that the linearization process described so far happens in frequency domain. Therefore, the convolution process that is rather complex in time domain is now reduced to simple multiplication of the loudspeaker's frequency response with the corresponding Amplitude Error Function.

The result of amplitude linearization is shown in Fig. 6. The resulting SPL (A) is now flat within the frequency range of interest, and the phase response (B) is almost smooth within the same frequency range.

At this point, you should examine whether you are making any real progress in time domain, as you are making it in the frequency domain. Look at the square wave being passed through the new, amplitude-linearized loudspeaker system (Fig. 7).

The news is good—you are making visible progress. The square wave after being passed through the amplitude-linearized loudspeaker system has now lost almost all ringing characteristics. It's still not a typical flat-top square wave, but you have already removed much of the imperfections.

Inspecting the non-flat shape of the square wave leaving the loudspeaker, you can suspect that, still, not all sinusoidal components of the square wave recombine correctly. Otherwise, the square wave would be flat already. Because you have taken care of maintaining the amplitudes of those sine waves by linearizing the frequency response of the loudspeaker, then the only remaining parameter that is still causing recombination errors must be the phase of the system.

Indeed, the phase response (B Fig. 6) is smooth, but not flat. If you could somehow “reverse” the phase response, you stand a good chance that combining the reversed phase with the loudspeaker's own phase delay will yield a flat phase response of the system.

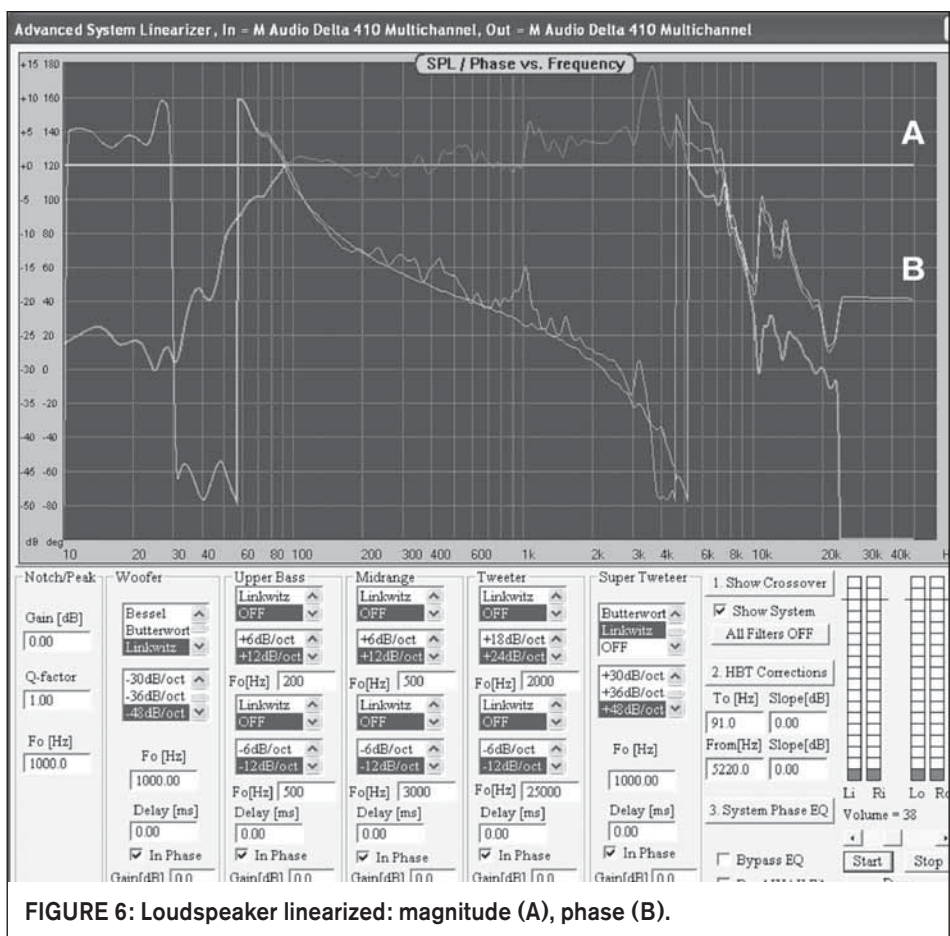


FIGURE 6: Loudspeaker linearized: magnitude (A), phase (B).

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INVERTING SYSTEM PHASE

A technique that allows you to reverse phase response of an LTI system is based on time-inverting its impulse

response. In practical terms, you need to reverse time scale of the impulse response. Typical impulse response described in time domain starts with a

high peak, followed by its decaying tail.

Imagine reversing time scale—it would be like the start of the impulse was the tail end of it, and then you arrive at the peak. If you perform this operation as convolution, interestingly, the phase response becomes its own mirror image around zero degrees. And that's exactly what you are looking for. In addition, if you are using FIR (Finite Impulse Response) techniques to accomplish the convolution, the amplitude response of the process can be decoupled from the inverted phase response and then forced to become flat.

You have now produced a perfect phase-reversal device with flat amplitude response—System Inverse Phase Function (Fig. 8). Remember, flat amplitude response requirement is important here, because at this stage, you do not want any more amplitude corrections. You have done this already in the previous stage, using the Amplitude Error Function.

To summarize, you now have two transfer functions:

- *Amplitude Error Function*—produced by inverted HBT. This is used to linearize the system's SPL and remove ripples from phase response within the *SPL frequency range of interest*.
- *System Inverse Phase Function*—this is used to nullify the whole system phase response.

Now apply these two corrective functions, in order, to the loudspeaker frequency response. In other words, multiply the loudspeakers' transfer function first, by the *Amplitude Error Function*, then take the result and multiply it by the *System Inverse Phase Function*. The function that you obtain in the end is the final transfer function of the fully equalized system. You can now try to pass your test square wave through such system and observe the result.

CONCLUSION

It is clearly evident that the proposed two-stage equalization technique brings about very good results. The resulting outgoing square wave is almost per-

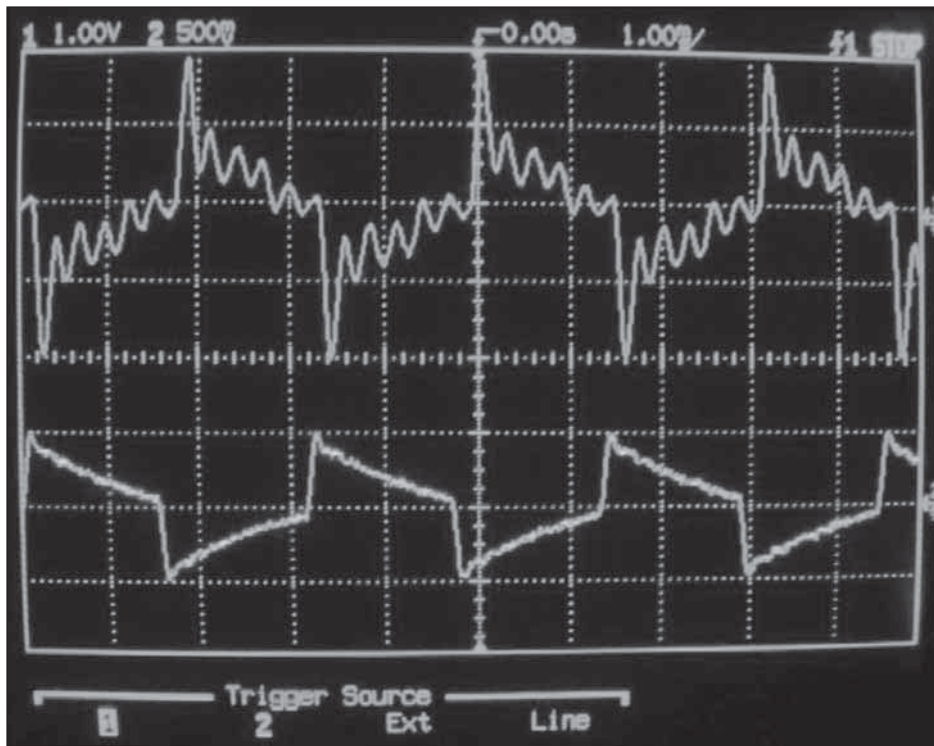


FIGURE 7: Square wave passed through amplitude linearizing system. Trace 1—loudspeaker alone, Trace 2—SPL equalized loudspeaker.

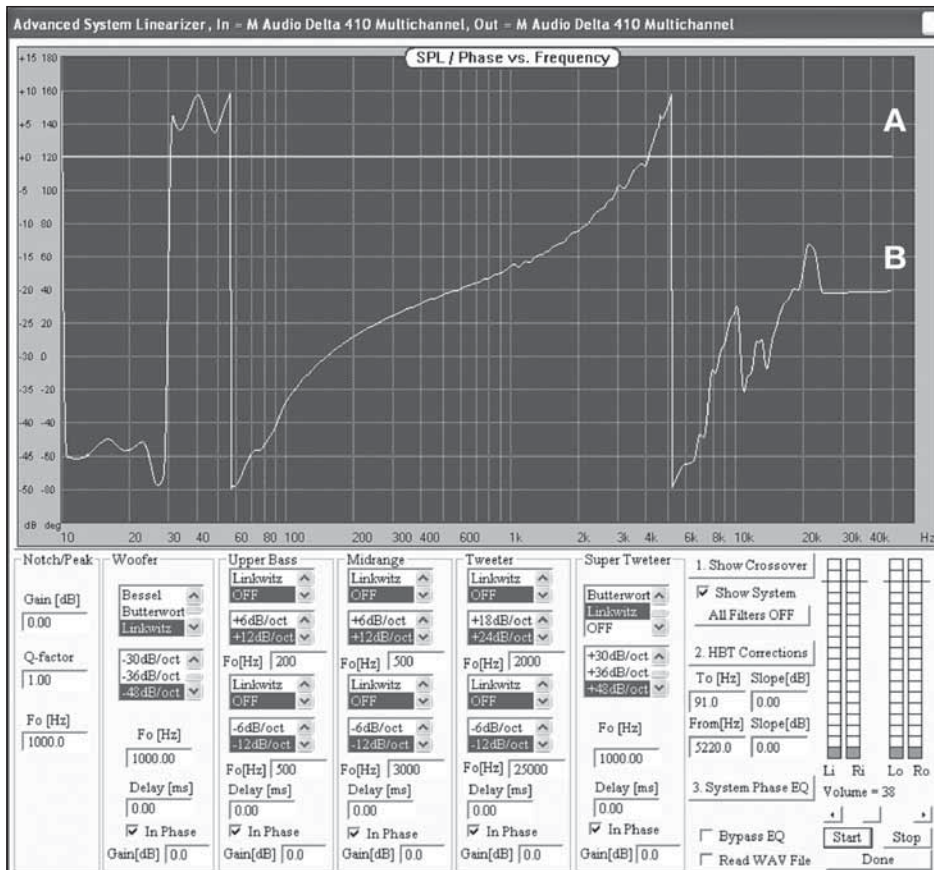


FIGURE 8: System inverse phase function: magnitude (A), phase (B).

fectly recombined from individual sine waves constituting the input square wave. This would be a confirmation that the equalized system now has flat amplitude within the frequency range of interest, and zero phase response. This loudspeaker can now play square waves.

A second, equally important, conclusion comes from analyzing **Fig. 6**. The Amplitude Error Function makes the magnitude and phase of the loudspeaker smooth, but it does not change macro-characteristics of the phase response, or rather, it does not "unwrap" the phase. The importance of this is that you can equally apply the HBT-based Amplitude Error Function to smooth the magnitude and phase response of non-minimum phase systems, such as a multi-way loudspeaker system, complete with crossover.

Also, the System Inverse Phase Function inverts the phase of the complete system, as it was measured, and regardless of the trajectory of the phase response. Consequently, the whole two-stage equalization technique is fully applicable to multi-way loudspeaker systems.

The third important conclusion is almost automatic. Linear phase response of the complete system (in this case, zero phase response) results in constant group delay—also equal to zero—of the complete electroacoustical system. As I explained at the beginning of this article, I mounted the loudspeaker used for this example in a vented enclosure. Therefore, the measured phase response already included characteristics (or phase delay) of the vented box. Because the finally obtained phase response was totally flat, it shows that the equalization process automatically eliminated group delay associated with the enclosure.

Finally, I would like to thank John Kreskovsky of Music and Design for the inspiration and exchange of ideas in the development of this procedure. *ax*

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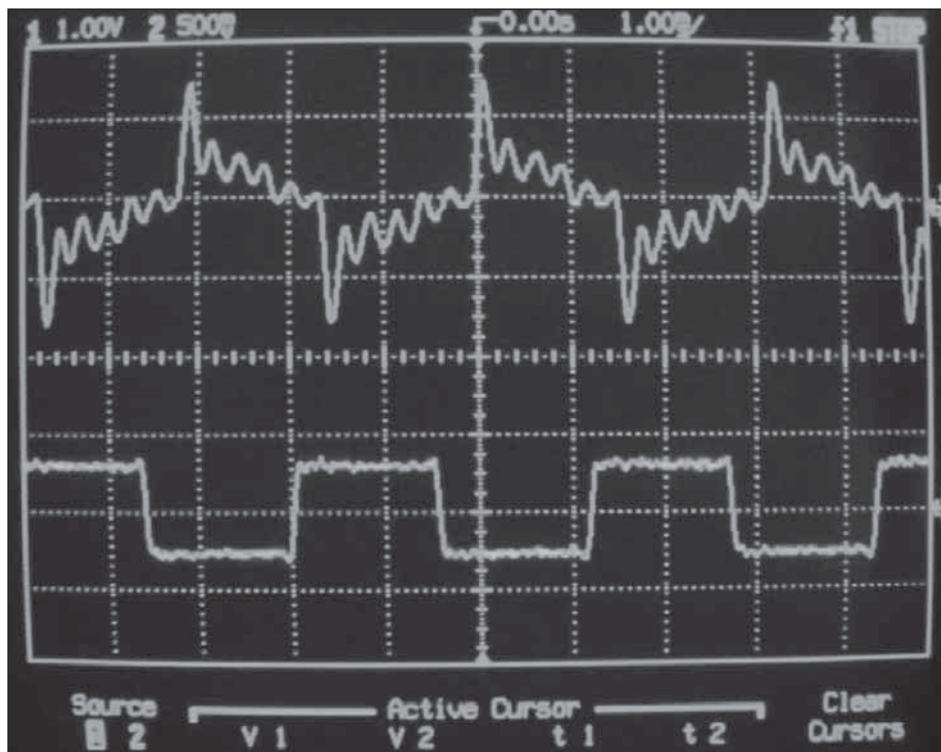


FIGURE 9: Top trace—300Hz square wave run through the loudspeaker alone. Bottom trace—the same square wave run through the loudspeaker and equalizer.

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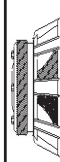
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The Optimized SRPP Amp, Part 2

With some theory behind us, it is time to really take the SRPP apart.

Last month, in part 1, I looked at the history of the SRPP (as quickly as I could), and covered the half- μ stage, which is often confused with the SRPP. I also hinted that, as a small power amplifier, there must be some optimum load impedance for the SRPP, one which gives (almost) the lowest distortion. It is now time to find what that load is, and along the way I will do my best to use as little mathematics as possible!

THE OPTIMIZED SRPP AGAIN

For audio purposes, you are most interested in making both valves in the SRPP contribute equally to the load

current, and also to maximize that current, and this is what I call the “optimized SRPP.” It is at or close to this condition that you obtain the lowest distortion.

Genuinely obtaining the absolute minimum distortion requires a distortion analyzer, because the critical values of resistance vary too much with manufacturing tolerances to be attacked on paper alone. However, even if you don’t have such test equipment, the values calculated using the formulas given here will always bring you close to the ideal, and will err on the safe side of less distortion rather than more. All you need is the HT voltage (i.e., the power supply

voltage or B+), and also the published figures for the internal anode resistance (plate resistance) r_a , and the amplification factor μ .

THE BYPASSED, OPTIMIZED SRPP

First examine the textbook version of the SRPP (Fig. 5A) when the lower bias resistor R_{k1} is bypassed by a large capacitor. From basic theory you know the quiescent current through a triode is:

$$I_o = \frac{HT}{r_s} + gm.v_{gk}$$

where:

gm = the transconductance of the triode

v_{gk} = the grid-to-cathode or bias voltage

For a cathode-biased stage, v_{gk} is determined by $-I_o R_k$, giving:

$$I_o = \frac{HT}{r_a} + gm(-I_o R_k)$$

Solving for I_o , you obtain:

$$I_o = \frac{HT}{r_a(gm.R_k + 1)}$$

And since $r_a gm = \mu$, this may be written as:

$$I_o = \frac{HT}{r_a + \mu R_k}$$

The simple SRPP (without the additional resistor R_a in Fig. 5) is two such triode amplifiers connected in series, so the denominator of the above is doubled to:

$$I_o = \frac{HT}{2r_a + 2\mu R_k} \dots\dots\dots [1]$$

The circuit is balanced when the quiescent current is half the maximum or peak current which could ever flow through the triodes. If you split the circuit into its two halves as in Fig. 5B, then you can see more easily what the maximum peak currents are. If the HT

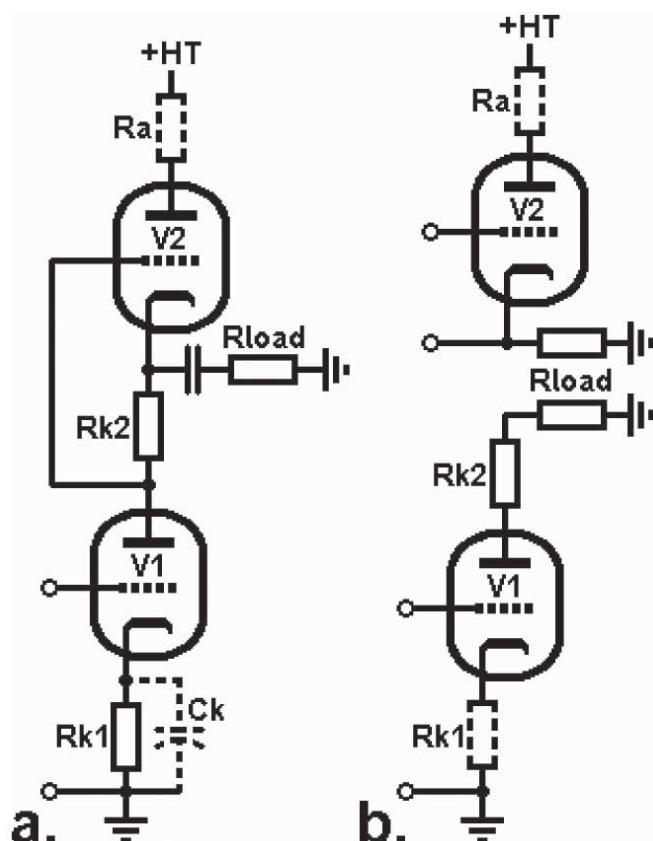


FIGURE 5: The textbook SRPP amplifier (A). Breaking it in two to see more clearly the resistances in the signal path (B).

is shared equally between the two triodes, then for the lower triode, assuming Rk1 is bypassed, the maximum current is $\frac{1}{2}HT/(ra + R_{load} + Rk2)$, while for the upper triode it is $\frac{1}{2}HT/(ra + R_{load})$. Because they are not quite balanced, you should add the additional resistor Ra, equal to Rk2, in series with the upper triode as shown, in order to equalize these currents. In other words, adding Ra equalizes the effective transconductance of the two devices (this was also included in Artzt's patent).

If you say that $Rk1 = Rk2 = Ra = Rk$, the peak current for either valve becomes:

$$I_{pk} = \frac{HT}{2(ra + R_{load} + Rk)}$$

So for proper balance the quiescent current must be half the peak current value:

$$I_o = \frac{HT}{4(ra + R_{load} + Rk)} \dots\dots\dots [II]$$

How can you force the quiescent current to equal this ideal value, for a given load impedance? There must be an optimum value of bias resistance Rk which will achieve this. This value is easily found by equating I and II (and remembering to add the extra resistor Ra). Thus you find the optimum value for the bias resistors:

$$\begin{aligned} 4(ra + R_{load} + Rk) &= 2ra + 2\mu Rk + Rk \\ 2ra + 2R_{load} &= ra + \mu Rk - 1.5Rk \\ ra + 2R_{load} &= (\mu - 1.5)Rk \end{aligned}$$

$$Rk_{optimum} = \frac{2R_{load} + ra}{\mu - 1.5}$$

Or, if you prefer to know the optimum load for a given Rk:

$$R_{load} = \frac{Rk(\mu - 1.5) - ra}{2}$$

However, with high- μ valves Rk will probably be very small compared to r_a , so the imbalance which occurs due to leaving out Ra is quite small. If you do choose to omit Ra, then the above can be simplified to:

$$Rk_{optimum} \approx \frac{2R_{load} + ra}{\mu} = \frac{2R_{load}}{\mu} + \frac{1}{g_m}$$

with only a modest increase in distortion¹. Applying Thévenin's theorem (which I won't prove here), you find the voltage gain of the bypassed SRPP:

$$A_{(Rk1bypassed)} = -\mu \frac{R_{load}(ra_2 + Ra + \mu_2 Rk)}{(ra_1 + Rk)(ra_2 + Ra) + R_{load}[ra_1 + ra_2 + Ra + Rk(\mu_2 + 1)]}$$

But for the optimized circuit where both triodes are identical and $Rk2 = Ra = Rk$, this simplifies to:

$$A_{(Rk1bypassed)} = -\mu \frac{R_{load}[ra + Rk(\mu + 1)]}{(ra + Rk)^2 + R_{load}[2ra + Rk(\mu + 1)]}$$

The output impedance is:

$$Z_{out(Rk1bypassed)} = \frac{(ra_2 + Ra)(ra_1 + Rk)}{ra_1 + ra_2 + Ra + Rk(\mu_2 + 1)}$$

which again simplifies to:

$$Z_{out(Rk1bypassed)} = \frac{(ra + Rk)^2}{2ra + Rk(\mu + 2)}$$

From the foregoing you see that there is always an optimum value for Rk for any value of R_{load} . If you change the load, you must change Rk too, or linearity will probably suffer (too many authors try to ignore the load when analyzing the circuit, which leads to quite misleading results).

It is therefore a mistake to use an SRPP as the output stage or line-driver of a preamp, for example, because you

can't be sure what it will be plugged into and hence what R_{load} might be. It could be as high as $1M\Omega$ if it becomes part of a valve power amp, or as low as $10k\Omega$ with a transistor amp, and you can hardly optimize for every eventuality. You can only properly design the SRPP if you know what the load impedance actually is, and treat the whole thing as a unified system (remember, an SRPP with no load is not an SRPP at all!).

However, if $\mu \gg 1$, and $R_{load} \ll ra$, then $Rk_{optimum}$ approaches ra/μ or $1/g_m$, and variations in load impedance will have much less impact. Unusually, then, the SRPP is best suited to low impedance loads. It is fundamentally a "small power amplifier," and ideal applications include driving dynamic headphones, which vary from around 15Ω to 300Ω and require only a few milliwatts of power. One early example of a commercial audio application was the Philips AG2126 "Magic Box" record player², in which a pair of triode-connected UL84s

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were arranged as a bypassed SRPP to drive high impedance loudspeakers.

If you have optimized the SRPP properly, then you now know that the peak load current is equal to twice the quiescent current, and from $P = I^2R$ the maximum output power must be:

$$P_{\max} = R_{\text{load}} \left(\frac{HT}{\sqrt{2} \left(r_a + \mu R_k + \frac{R_a}{2} \right)} \right)^2$$

What may not be immediately obvious is that *absolute maximum power* from the SRPP obtains when the load impedance is equal to r_a , so, in theory, it should be possible not only to optimize the circuit resistances but also to choose the optimum triode for the job! All you need to do is find one whose r_a is equal to the impedance of what you wish to drive, and which can also handle the dissipation, of course. For example, if the load is 10k Ω , then an ECC82/12AU7 comes pretty close with $r_a \approx 10\text{k}\Omega$.

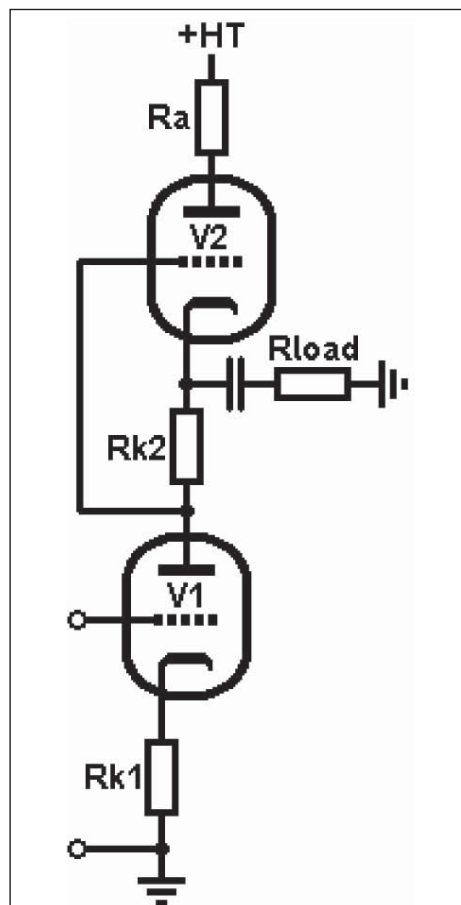


FIGURE 6: With this unbypassed version of the SRPP, $R_{k1} = R_{k2} = R_k$, and the optimum value is:

$$R_{k_{\text{optimum}}} = \frac{2R_{\text{load}} + r_a}{\mu - 1}$$

R_a should be twice this value.

LESS DISTORTION AND MORE POWER?

I hope the preceding section illustrates how the optimization process works. However, you are more likely to find SRPPs where all resistors are unbypassed, as in Fig. 6. Not only does this avoid the use of another large, electrolytic capacitor, which audiophiles universally hate, but it also reduces distortion and even facilitates more output power! The last point may seem counterintuitive, but will become clearer later.

The disadvantages of this arrangement are that the voltage gain and PSRR will be lower, and the output impedance will be much higher. On the other hand, because you are designing the circuit specifically to deliver power into a known fixed load impedance, the output impedance should be of only marginal concern.

Another reason you might not wish to bypass R_{k1} is that when it is bypassed, the even-harmonic distortion of both triodes will largely cancel out, leaving only low-level odd harmonics, which is what you would expect from any balanced push-pull amplifier. However, by leaving R_{k1} unbypassed, current feedback will greatly reduce the distortion generated in the lower triode as compared with the upper triode. The total distortion of the circuit will then be dominated by the second-harmonic distortion of the upper triode, retaining that “warmth of tone” that you expect from triodes.

THE UNBYPASSED, OPTIMIZED SRPP

If R_{k1} is unbypassed, then R_a should be made equal to twice R_{k2} for proper balance, although, again, the error is small with high- μ triodes. If the resistor is included, though, then the optimum value for R_k becomes:

$$R_{k_{\text{optimum}}} = \frac{2R_{\text{load}} + r_a}{\mu - 1}$$

And the quiescent current will be:

$$I_o = \frac{HT}{2[r_a + R_k(\mu + 2)]}$$

The voltage gain is:

$$A_{(R_{k1} \text{ unbypassed})} = -\mu_1 \cdot \frac{R_{\text{load}}(r_a + R_k(\mu_1 + 1))}{[r_a + R_{k2} + R_{k1}(\mu_1 + 1)](r_a + R_k(\mu_1 + 1)) + R_{\text{load}}[r_a + r_a + R_k(\mu_1 + 1) + R_{k2}(\mu_2 + 1)]}$$

But if you don't like the look of that then for the optimized circuit it simplifies to:

$$A_{(R_{k1} \text{ unbypassed})} = -\mu_1 \cdot \frac{R_{\text{load}}[r_a + R_k(\mu_1 + 1)]}{[r_a + R_k(\mu_1 + 1)](r_a + R_k(\mu_1 + 1)) + R_{\text{load}}[2r_a + 2R_k(\mu_1 + 1)]}$$

The output impedance will be much higher than for Fig. 5:

$$Z_{\text{out}}(R_{k1} \text{ unbypassed}) = \frac{(r_a + 2R_k)[r_a + R_k(\mu + 2)]}{2r_a + 2R_k(\mu + 2)}$$

And the maximum output power is:

$$P_{\max} = R_{\text{load}} \left(\frac{HT}{\sqrt{2} [r_a + R_k(\mu + 1)]} \right)^2$$

The above equation still does not give anything away, but when you observe that the optimum value for R_k is *less* than for the bypassed version, you will find that the available output power is actually greater! However, before you deride the bypassed version for being a weakling, note that the difference in power is quite negligible for triodes with μ greater than about 10.

THE SRPP+

In some cases you will find that the optimum value of R_k is too small to bias the valves properly without causing them to overheat. If you can't increase the load impedance and you don't want to use a more powerful pair of valves, you can instead bias the valves to any value of current you like, and restore balance by connecting the load to a tapping point on R_{k2} instead. Or, in other words, split R_{k2} into two parts, R_1 and R_2 . This gives the circuit in Fig. 7, which was rediscovered recently by Broskie³ and to which he has given the name SRPP+.

The supply voltage and quiescent current can be chosen first, according to the desired dissipation in the valves (which is approximately $P = HT/2 \times I_o$). Rather than bore you with the same mathematical treatment again, I will quickly consider only the unbypassed version of the circuit using high- μ triodes, which allows some simplifications to be made.

Having decided what quiescent current I_o is allowable, the necessary value for R_k to bias the triodes to this current is:

$$R_k = \frac{HT / 2I_o - r_a}{\mu}$$

$$R_{\text{load}}(r_a + R_k(\mu_1 + 1))$$

And $R_1 + R_2$ must also equal this value. The purpose of R_1 is to deliberately degenerate the transconductance of the upper

triode so that it becomes equal to $1/R_2$, at which point the maximum peak current in the upper triode will (more or less) match the lower triode, and will be equal to twice the quiescent current, just like the other optimized SRPPs. The value of R_1 is:

$$R_1 = \frac{HT / 4I_o - r_a - R_{load}}{\mu}$$

And R_2 is simply $R_k - R_1$, or if you prefer to find R_2 directly:

$$R_2 = \frac{HT / 4I_o + R_{load}}{\mu}$$

which is slightly simpler than the previous equation. If you find yourselves with a negative value for either of these resistors, then it indicates that the load impedance is simply too large for the circuit to work. This demonstrates yet again that the SRPP actually prefers low-impedance loads!

Strictly for AC balance, you should add an extra current-balancing resistor R_a , having a value of $R_k + R_2 - R_1$. However, the error caused by omitting this extra resistor is even smaller than for the ordinary SRPP. This is fortunate, because leaving it out considerably simplifies the math. Unfortunately, the exact expressions for gain and output impedance of this circuit

are painfully long, and liable to make half the readership faint at the very sight of the SRPP+. Therefore, I shall be diplomatic and say that the gain, output impedance, and output power are *very nearly* the same as for the ordinary, unbypassed SRPP, with the same value R_k and R_{load} .

CONCLUSION

Although I have not been exhaustive, I hope this article has helped to de-mystify the operation of the SRPP for some readers, and illustrated what the SRPP does best. It is a genuine push-pull amplifier, and is best suited to small power applications, and will drive surprisingly heavy loads. Dynamic headphones (normally around 15Ω to 300Ω) are one ideal application, and a pair of ECC82/12AU7, ECC88/6DJ8, or 6SN7 are more than capable of making a good SRPP headphone amplifier which will drive most types of headphones—even cheap, low impedance earbuds!

The textbook version of the SRPP, where R_{k1} is bypassed, offers about twice the gain and half the output impedance compared to the equivalent unbypassed version. But the available output power is actually slightly less. So unless you really need the extra gain, most designers will choose the unbypassed version. Not only does it offer less distortion, but the higher output impedance will allow the use of a smaller output coupling capacitor.

For example, an ECC88/6DJ8 driving a 15Ω headphone would have an output impedance of about 910Ω bypassed, or 1380Ω unbypassed. For a cutoff frequency of 20Hz, the bypassed version would need a $9\mu F$ coupling capacitor, whereas the unbypassed version would need only $6\mu F$, making it much easier to use a good-quality, non-electrolytic capacitor. Yet from the same 200V power supply both would deliver a maximum of 10mW, with surprisingly good linearity!

This illustrates rather nicely how the output impedance of the SRPP is largely incidental, because the whole circuit is designed with a specific load impedance in mind. (By the way, if you actually choose to build such a circuit, the unbypassed version would need $R_k \approx 82\Omega$, giving a quiescent current of 19mA. The gain would be about 0.2, so you would need an input voltage of about 2V RMS for full output.)

The SRPP is not an all-purpose line driver, because in such cases the load im-

pedance could vary over a very wide range. A load impedance which is somewhat higher than the design value might not be too bad, but one which is lower would be disastrous. A cathode follower would be a much more sensible choice for this position.

On the other hand, if all you need is voltage gain, and the load impedance is fairly high (a few hundred kilohms or more, say), then the half- μ stage is a much more sensible choice than the SRPP, and offers extremely consistent results. It could make an ideal input stage for an MM phono stage, for example, where predictable gain and input capacitance are essential. And if this article has done nothing else, then I hope that it will at least make you look a little closer before giving just any totem-pole stage the grand title of SRPP! *aX*

REFERENCES

1. This is the same formula as presented in Millman, J. & Taub H (1956), *Pulse and Digital Circuits*, McGraw-Hill, New York, p. 100.
2. (1956) Radio Show Review, *Wireless World* (October), p. 475-6.
3. <http://www.tubecad.com/2009/09/blog0172.htm>.

ADDENDUM

In Part 1 of Merlin Blencowe's article, we inadvertently repeated *Figure 3* as *Figure 4*. The correct *Figure 4* appears here.—Eds.

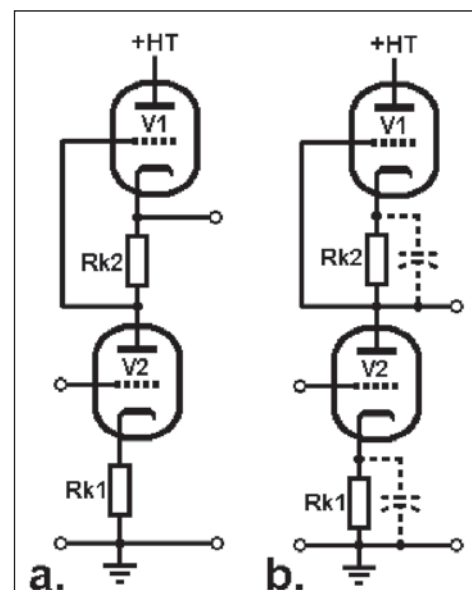


FIGURE 4: With an infinite load the circuit is not an SRPP at all, but an actively loaded triode (A). By moving the output to the anode of V1, you obtain the half- μ stage (B). Bypassing both resistors reduces the output impedance, though distortion will be greater.

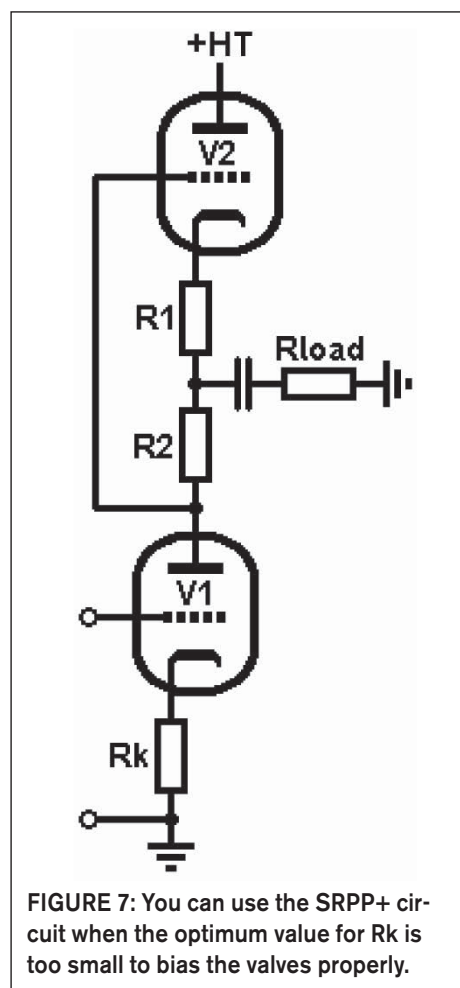


FIGURE 7: You can use the SRPP+ circuit when the optimum value for R_k is too small to bias the valves properly.

Bryston BCD-1 CD Player

By James T. Frane

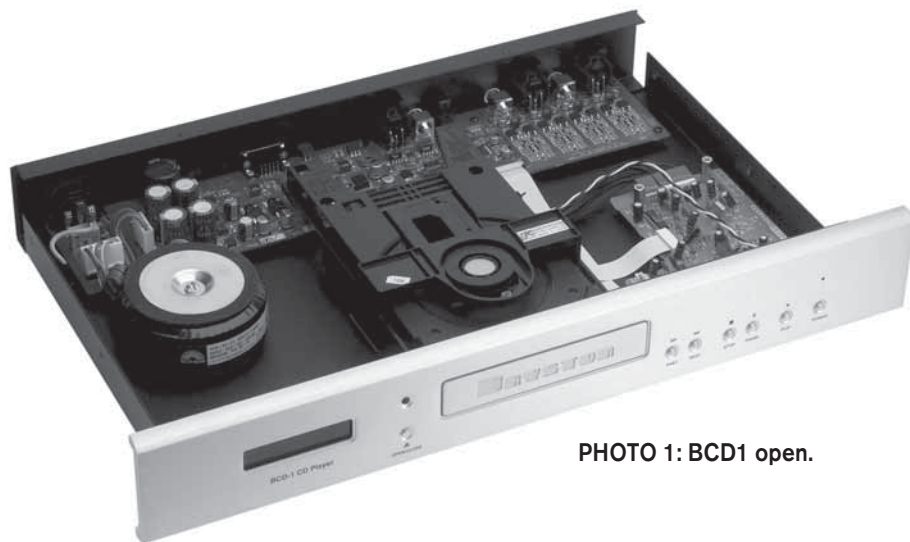


PHOTO 1: BCD1 open.

I recently purchased Bryston's BCD-1 compact disc player to replace a decade-old (but still very good) Sony CD player. A few years back I reviewed some Bryston amplifiers and was quite taken with the best music I had heard from my system. After the reviews were published, I purchased a Bryston preamp and power

extends out to hold the disc on four rubber pads so the bottom disc surface is protected. "BRYSTON" is cut into the drawer front. On the last third of the panel there is a row of buttons for PREVIOUS, NEXT, STOP, PAUSE, PLAY, and POWER. All of these buttons are duplicated on the remote control.

“ [The Bryston BCD-1's] reproduction of music through my system exceeds the performance of any other CD player I have heard. ”

amp and have enjoyed their excellent sound. When I learned of Bryston's CD player and researched the available literature, I purchased one for my system.

The BCD-1 CD player is available in black or silver finish and with either a 17" or 19" wide front panel. I opted for black with the 17" panel to best fit my system. Starting from the left side of the front panel, a window provides a status message, such as NO DISC, TRACK and PLAY or PAUSE, as well as the number of seconds the track is playing. The lighted display is in easy-to-read bright green letters and numbers. To the right of the window is an infrared sensor for the remote control, and directly below is the open/close button for the drawer.

Further to the right is the drawer that

There are several options to connect the BCD-1 to a stereo system preamplifier. You have two analog output choices:

- balanced (XLR)
- single-ended (RCA) connections

and three digital output choices:

- TOSLINK (optical fiber connection)
- AES/EBU (officially AES3 developed by the Audio Engineering Society and European Broadcasting Union for digital audio signals)
- S/PDIF version of AES3 that uses less expensive hardware

I connected the BCD-1 to the preamp with 1m long XLR cables.

LISTENING

I have written about several CD players auditioned in my system, includ-

ing some in the \$3,000+ price range. None of these sounded consistently better than my older Sony, and some were not as good. Generally the sound of my best records was a little better than any of my CD collection—that is, until the Bryston BCD-1 arrived. Its reproduction of music through my system exceeds the performance of any other CD player I have heard.

For the first time, CDs sound as good or better than records of the same music. With a well-recorded CD the BCD-1 provides a three-dimensional soundstage as well as a correct and consistent CD playback. The key to this good performance is Bryston's quality design and construction that pays attention to the Red Book technical details of standards for writing digital music to discs. Red Book Audio Specifications are the physical parameters and properties of the CD, including deviations and error rate correction, and the form of digital audio encoding. Bryston engineered the BCD-1 to take full advantage of these standards in the reading and playback of a disc.

The following are but a few examples of the dozens of CDs I played on the BCD-1:

Willie Nelson's "Georgia On My Mind" from Willie Nelson 16 Biggest Hits (Columbia/Legacy CK 69322) was center stage with good dimension and sound. In addition, the first guitar string pick, which is usually most noticeable on the LP, was very clear.

Roger Waters' "Late Home Tonight, Parts I and II" from his Amused to Death album (Columbia CK47127), provided virtually surround sound. Not a CD for everyone, but beautifully performed and recorded.

Paul Desmond's "Skylark" from the album of the same name (CBS Associated ZK 44170) placed him squarely at center stage with drum set to the right, piano to the left, and guitar center

stage behind Desmond.

Julie London's "Cry Me A River" from The Best of Julie London (Capitol/Rhino R2 70737) was like a live performance. The strings a bit behind her were clear, precise, and full-bodied with feeling. Stage depth was realistic, and Julie's voice engaging.

Cream's "Wrapping Paper" from The Very Best of Cream album (Polydor 31452 3752-2) had good depth, with each voice and instrument clear and precise. The sound nicely portrayed the recording venue spaciousness.

"Jesu, Joy of Man's Desiring," by the Wind Group of the American Symphony Orchestra, conducted by Leopold Stokowski (Vanguard Classics OVC 8009) in 1966, had the orchestra spread in front of me with depth and the harmonious blending of instruments where appropriate. So easy to listen and be transported to the performance. A large number of 1950s and 60s music gained new life when played on the BCD-1.

OPERATION

Bryston literature states that the BCD-1 is a state-of-the-art Redbook CD player using completely discrete or separate analog Class-A proprietary Bryston circuits, a quality drive, and a 192K/24-bit crystal DAC (digital to analog converter). The BCD-1 also features discrete class-A output amplifiers and a 24 bit/192K over-sampling DAC. For the technically minded: Getting the digital side of the compact disc equation correct is essential for correct playback. Once the signal leaves the DAC, it is buffered and increased in strength by operational amplifiers.

In the BCD-1 CD player these amplifiers are constructed from discrete devices (individual transistors, resistors, and capacitors) rather than the integrated circuits of many CD players. Discrete devices allow circuit design that exactly matches the functions of the DAC as well as more powerful outputs from operational amplifiers. The heat from the output driver transistors

is separated from other devices for better overall performance. Bryston states that the advantage of an all-in-one box solution for a CD player (as opposed to a separate outboard DAC and drive) is the elimination of jitter (defined as mistiming of the digital signal).

They explain as follows: "For optimum performance the drive and DAC must use the same master clock. If the clock signal of the drive is not synchronized with the clock signal from the DAC then jitter develops. In external DACs the digital input must be re-

clocked in order to reduce the jitter. The timing of all those ones and zeros is of extreme importance. It isn't enough to get the bits right; those bits have to be converted back into music with the same timing reference as when the music was first digitized. These timing errors (jitter) are the bane of quality audio. With the Bryston CD player the master clock and the drive are synchronized perfectly to eliminate any possibility of jitter affecting the sound quality of the player."

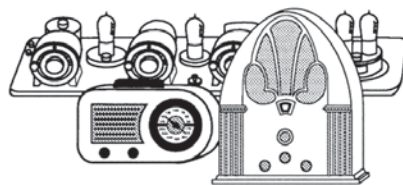
THE BR2 REMOTE CONTROL



PHOTO 2: BCD-1.

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The 2 × 6 × 7/8" (approximately) remote control that comes with the BCD-1 is matte black or silver to match the CD player finish. It is solidly built of machined aluminum. There are 30 small buttons to operate the various functions of the player. The remote is powered by two AAA batteries, which you can change by removing the four screws on the back of the unit.

A motion sensor in the remote backlights the buttons with blue light when you push them. This is a welcome feature when listening in a dim or darkened room, but you can disable it by entering a three-digit code on the keypad. The BR2 remote control provides a variety of functions, including the ability to control various other Bryston electronics. I programmed it to adjust my Bryston preamp volume just as the remote that came with the preamp does, but this unit offers a finer degree of control and eliminates the need for two remotes.

- The top two buttons on the remote are "code," which allows programming the remote when used in conjunction with other buttons (as shown below) and "Power," which is used to turn the player on and off from a distance.
- The other 12 buttons in the top half of the remote provide the possibilities to program (using one of the three

digit sequences listed in the manual) a Bryston preamp or other Bryston components so they respond to commands from this remote. Programming is quite straightforward. You hold down the "code" button until a red LED flashes and then enter the listed three-digit sequence (from the manual) for the desired program.

- The remaining 16 buttons provide a variety of options to control the CD player from a distance: play, including random, repeat, next, back (to replay part of a cut), previous (to go back to the preceding cut), forward, eject, and so on. The remote can be in many locations and orientations relative to the player and still be operable—a relief from those remote controls that require "spot-on" aiming.

When you insert a new disc, there is a few seconds delay as the BCD-1 processes the information on the disc. From then to the end of the disc, even when the unit is set to random play, each cut smoothly succeeds the previous.

TECHNICAL INFO

The DAC integrated circuit (chip) provides the conversion of the digital signal to the analog domain. The chip used in the BCD-1 is a Crystal CS4398, which is a hybrid multi-bit delta-sigma DAC. This is a rather tongue-twisting name that refers to an advanced generation chip with a combination of several different methods to optimize the conversion process with a process to over-sample the digital input 128 times.

Over-sampling is re-reading (2x, 4x, 8x, and so on) to produce a new sampling frequency. The new samples are then run through an interpolation filter to produce a more analog-like waveform. The output of this process is a very sensitive analog signal, and it is critical that the timing is very closely controlled by a low jitter clock.

DAC chips also require a very clean digital power supply if they are to function at their optimum. Noise on the digital supply may cause added jitter,



PHOTO 3:
BCD-1 remote.

noise, and distortion. Incorrect circuit trace routing of the digital power supply or ground may introduce digital noise into the analog circuits. This digital power supply is provided from a separate closely regulated and filtered source.

The DAC also requires a high-quality analog power source because the analog signal is at its lowest magnitude when it enters and leaves the DAC. As a result, any added noise or distortion will be greatly amplified by later stages. A separate, heavily regulated and filtered power supply is provided.

Discrete devices allow specific matching of important characteristics such as input and output impedances based on the in-circuit requirements. They can be tested very closely to meet defined tolerances and can be matched against each other when a desired performance design is critical. Operational amplifiers can be designed to closely match their power source for additional reduction of distortion and noise.

Bryston states they do sorting and grading with DAC chips to a fine level to achieve superior sound quality. They state they hand-assemble and individually test each product they manufacture and exclusively use only 1% metal-film resistors, polystyrene capacitors, and hand-selected and matched transistors in order to reduce noise and distortion to the absolute minimum.

MANUALS

Brief owner's manuals are provided for the BCD-1 CD player and another for the BR2 remote control.

Main operating controls and their functions on the remote are:

PREVIOUS: Move to the previous track.

NEXT: Move to the next track.

STOP: Stop playing disc. Pressing "play" again will return to the beginning of the first track.

PAUSE: Stop playing disc while retaining the current position. Press Pause again to resume play.

PLAY: Play the disc. Pressing Play when

the disc is already playing will cause the BCD-1 to replay the disc from the beginning of the first track.

Power on-off button: Toggles the power on and off. This button will not function if the External Trigger function has been activated using code 241.

Specifications are given as:

- Frequency response: 20Hz-20kHz, .2dB
- Signal to noise: 115dB unweighted
- THD plus noise: .002%
- Jitter is negligible (below the residual of the Audio Precision AP2700 test gear)
- Output level: 2.3V unbalanced, 4.6V balanced
- Shipping weight: 18lbs/8.2kg

- Dimensions: 17 or 19" wide/11.25" deep/3.125" high (43.2 or 48.3cm w/28.6cm d/7.9cm h)

Associated equipment:

Bryston BP25 preamp and SST 4B power amplifier, Thiel CS 2.4 speakers, Thorens TD320 Mk III turntable with Grado Reference Platinum phono cartridges, Grado PH-1 phono preamp, Sunfire HRS-10 Subwoofer, Sony CDP-XA-2ES CD player, Magnum Dynalab MD100 FM tuner, Kimber 8TC speaker cables, Bryston balanced interconnect cables (preamp to amp and BCD-1 CD player to preamp), and Vampire and WireWorld unbalanced (RCA) interconnects. **rr**



PHOTO 4: BCD-1 back.

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CORRECTION

In the article "Amplifier Comparison Using Oscilloscope Waveform Plots" (5/10 *aX*, p. 20), the captions for Figs. 1 and 2 showing amp performance were inadvertently switched. Figure 1 shows the solid-state results, while Figure 2 is the tube plot.—Eds.

POWER AMP MOD

I am reaching out to Kit Ryan for help and advice regarding his article, "A Power Supply Regulator for the Adcom GFA-555" (*Audio Amateur* 4/89). I recently came across this article and am very intrigued and interested to transform my trusted GFA-555 into

what you describe as "truly first class equipment." The Toshiba pass transistor pairs 2SA1095/2SC2565 are now virtually impossible to get from a genuine source.

I was wondering whether over the years you had revised the design with newer devices or looked at possible substitutes, perhaps from the newer OnSemi family? I realize the limitations of the 1" × 1" aluminum angle also influenced your decision in selecting the flat Toshiba rather than the T-03. I am eager to build this mod and would really appreciate your advice in this matter.

Mayank Bhargava
Falls Church, VA

Kit Ryan responds:

Believe it or not, I've been using my original Adcom 555 with these power supply mods for the last 20 years. After trying a couple of other amps, I still haven't been impressed enough to change them out. Doing mods to other equipment in the interim has convinced me that the power supply is probably at least as important to good sound as the actual amplifier circuitry of much audio equipment.

I did a little research and agree that my original transistors are no longer available anywhere. ON Semiconductor (www.onsemi.com) does make some good-looking replacements, however. Take a look at the MJW3281A and its complement, MJW1302A. These have similar power, volt-

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are a little awkward and sometimes humorous, but the information is understandable.

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age, current, and safe operating area characteristics as the original transistors, plus they will also mount flat against an aluminum heatsink. There are slightly lower power versions—MJL3281A and MJL1302A—that might do, too. I think you can probably make either of these work. An added bonus is that the pin spacing and BCE configuration appear to be identical to the originals.

As with all power amplifier circuitry, be very careful with assembly and testing. Even a small mistake can blow out a lot of expensive parts. I short-circuited the output once and it didn't survive, I had to replace the circuit. So don't do a destructive test on it. Under all other playing conditions, though (I've played it to beyond clipping), there've been no problems.

PHONO PHASE

Over the last few years there's been sporadic discussion of the use of digital processing to add the RIAA phono playback curve to a recording that was made through a flat preamp. The concern has been that because some audio processing software uses "linear-phase" filters, which do not add phase shift as they alter amplitude response, such a method would fail to properly compensate for the phase shifts introduced when the record was cut and the RIAA pre-emphasis was added.

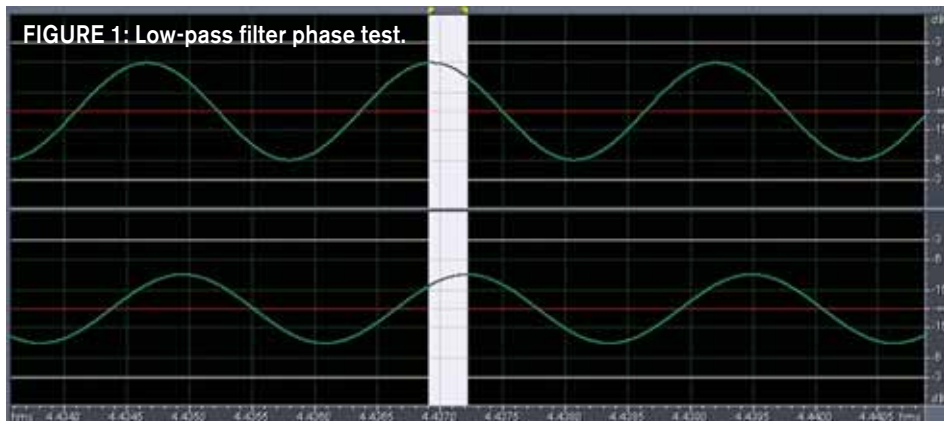
The results of querying manufacturers of software about which type of filtering they use have been inconclusive. Re-reading some of these discussions, it occurred to me that this question could be settled pretty easily by experiment—so I tried it.

I created a stereo file with identical sine waves in both channels, using Adobe Audition, then processed the right channel with a low-pass 6dB/octave digital filter with a -3dB point equal to the frequency of the sine wave.

The result is shown in **Fig. 1**. The low-pass filtered signal has been reduced by 3dB, and shows a textbook 45° phase lag, just as it would if it had been processed by an analog filter. I tried the same experiment, using the audio restoration program DC-SIX (the current version is DC-SEVEN), and got the same results.

For these two programs, it seems that the processing is indeed minimum-phase rather than linear-phase, which means they could be used to generate an RIAA curve, or any other playback

FIGURE 1: Low-pass filter phase test.



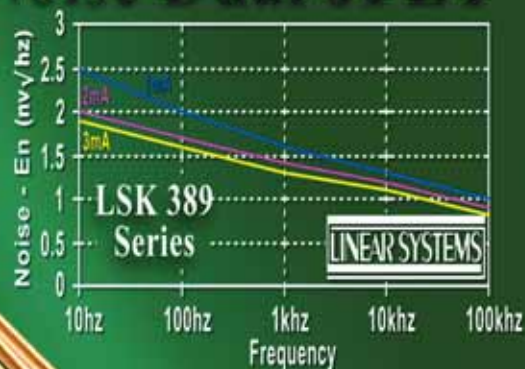
compensation curve needed for playing non-RIAA records, which will accurately preserve the phase relationships recorded onto the discs.

*Paul J. Stamler
St. Louis, MO*

Loyal reader and author, Atto Rinaldo of Tambre in the northeastern part of Italy, sent us this photo of his collection of Audio Amateur publications—bound in book format—and starting with the first issue of *Glass Audio* (1998),--Eds.



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CABLE HUM

I have comments on the article on magnetic shielding ("Balanced Cable Hum: Cause and Cure," May '09 *aX*, p. 34). I chase electromagnetic interference (EMI) gremlins for a job. I got into this with physics and electrical engineering degrees and have done it since way back.

I attach a 1956 Boeing aircraft company study of magnetic field coupling in twisted pairs (**Fig. 1**). I suspect that this was done for instrumentation at the time that was analog, or for audio distribution. I have referred to it when looking at EMI problems.

There is a company that makes shielded cable for instrumentation that uses mu metal braid for shielding and uses four wires braided together as counter twisted pairs to reduce magnetic pickup. It is really difficult stuff to work with as the braid takes a long time to comb out. I had tried this on 450V DC 600A magnet drive amplifier current shunt sense wires.

Even with 4 wire 2 Ga cables braided and the braided shielded as shown, shielding was not adequate. Hmmmm. The technique would probably be adequate for low-level audio. If you are interested, I can dig through the catalog file and scan in a datasheet.

Owen Gallagher
Hudson, Mass.

PDRS

I've been following Keith Levkoff's series on Personal Digital Recorders (March and April '10), and I would like to point out one more model with some distinguishing characteristics.

Some years back, I acquired my dream recording system consisting of a pair of AKG C313B-ULS microphones, a Tascam DA-P1 DAT recorder, Aardvark interface for SPDIF transfer to my computer, CoolEdit (now reincarnated as Adobe Audition) software, and a Plextor CD burner. The ergonomic reliability of the Tascam professional recorder was wonderful; it relieved me of the serial copy management issue, and it recorded at 44.1 to avoid conversions for CD, unlike my first-generation Sony DAT that I was renting from the repair shop. Although it served me well, the Tascam mike preamps were lacking in headroom and were too noisy (-61dB), yielding results not much quieter than a good analog tape machine. The real-time transfer to my computer was getting tiresome, and the defunct Aardvark interface is not supported for my latest computer. My search for something better led me to the M-Audio MicroTrack II (www.m-audio.com).

I've been using the MicroTrack II to record symphony and choral concerts for over a year. Although I could afford

RELATIVE SUSCEPTIBILITY OF CIRCUITS TO
MAGNETIC-FIELD COUPLING AT 40CHz
PER BOEING STUDY D-183C6,1956

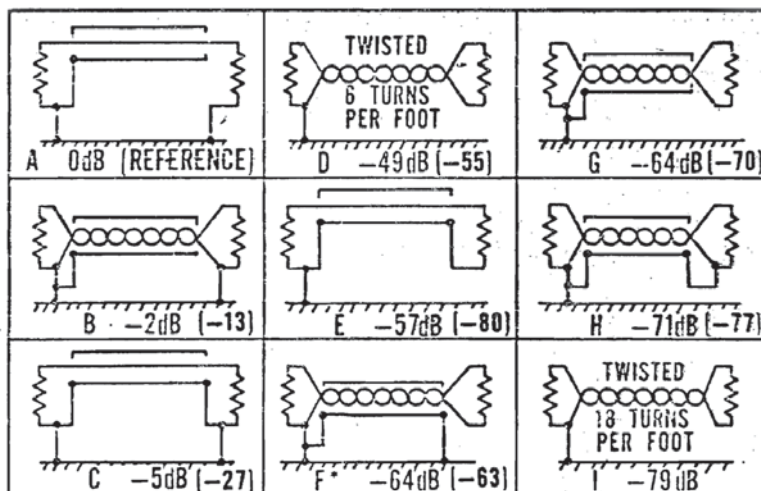


FIGURE 1: *Preferred circuit for high frequencies. (I) Indicates test results at 50kHz. Values given are for circuits 2" above ground plane, but are about the same for other distances from ground plane.

the cost of a more professional recorder, this one seduced me with its bargain price and outstanding performance that was confirmed by the on-line reviewers. (Recently, I've seen it on sale for well under \$200.)

Unlike most PDRs, it supports my balanced line condenser microphones with 49V phantom power and 90dB of headroom—a low enough noise floor to fully exploit 16-bit recording without the expense and hassle of a separate preamp. (The recorder includes a 24/96 mode and SPDIF input, but I don't see the value if the recording is going on to CD with minimal editing.) Its built-in limiter has saved my butt when the percussionist gets an adrenaline rush that wasn't there at rehearsal level checks—it triggers at -1dB and is virtually inaudible if not hit too hard. Some very affordable condenser microphones have come on the market in recent years, and combined with this recorder, they can make for a very affordable system of professional sound quality.

The good news and the bad news is the small size of the MicroTrack II. For seniors like me, a lighted magnifier is required support equipment for the concert hall. The supplied microphone provides adequate quality for casual recording but has little or no stereo separation. I chose this recorder mainly for the balanced and powered external mike capability, but that included some unanticipated problems.

Its small size dictates TRS jacks. Using them requires taping the recorder down to the desk. With such a small and light recorder, it is almost impossible to touch it without it moving, and the slightest motion can cause the TRS jacks to move introducing noise spikes at full clip due to the 48V phantom power going through the connections. (I suggest that the next model should use a more secure multi-pin connector with in-line XLR plugs on a dongle, which is what I have anyway to adapt to the microphones.)

The internal battery drains in about 20 minutes while powering my microphones, but I solved that with an external USB supply that uses disposable AA batteries. Most of the time, commercial power is available, so I just plug it in. I also needed to construct my own drop-

ping resistor box with blocking capacitors for dubbing from my stereo. There are no AUX inputs (only outputs), and the factory dropping plug passes the DC for electret microphones which caused problems dubbing from my system (speaker destroying transients when charging the coupling capacitors when changing sources).

I wish M-Audio would repack this performance in a professional case with easy-to-read displays, less scattered and crowded controls, XLR and no-DC auxiliary inputs, and space for disposable batteries that would last for a warm-up plus the full concert (or hot-swap against the built-in rechargeable) while using phantom power.

Dennis L. Green
d7green@yahoo.com

Keith Levkoff responds:

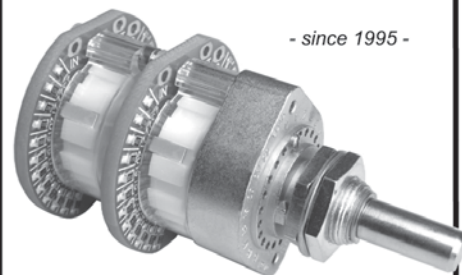
There are a whole bunch of different PDRs available—especially if you're willing to go up to the \$300-\$400 range—and many of them include all sorts of nice features (check out some of the comparison reviews at the links following part 2 of my article). I picked the Zoom H2 because it was the cheapest one which still offers a relatively complete feature set (I've seen them lately for \$140). Considering the nice big color screens I've seen lately on pocket TVs and other similar gadgets, PDRs are still a bit behind the curve there. I know a few people who have trouble reading those tiny displays—it would certainly make sense to fill the whole side with a nice big bright one—and how about a feature where the back light comes on when you touch the unit or tap the display gently?

You might check out the new Zoom H4n model (www.zoom.co.jp), which includes XLR connectors and 48V phantom power and claims a very nice battery life and a larger screen. (I haven't played with one yet.)

Along with the Zoom, I also have an Edirol R09-HR, whose screen, while no bigger than the one on the Zoom, is a lot brighter and easier to read, but I'm not sure that I can fairly say it justifies the higher price (about twice as much). If you looked at the Edirol R09 in the past, you should look again at the R09-HR—they've eliminated the weird battery door and screen and added 96/24 operation for about 50 bucks less (www.edirol.com). aX

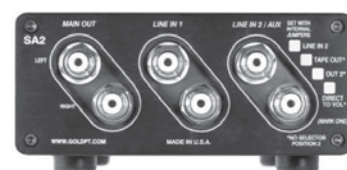
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CONTRIBUTORS

Andreas Schilloff ("A Three-Way Tri-Amp Center-Channel System," p. 5) lives in upstate New York with his wife Kristi and dog Sally. He has an engineering degree from Rochester Institute of Technology and works for a global EMS company. This is his second article for *audioXpress*.

Bohdan Raczynski ("A Loudspeaker That Can Play Square Waves?," p. 11) has a Master's Degree in Acoustics and is currently a director, running a small software company "Bodzio Software" in Melbourne, Australia. Born in Poland, he developed a passion for audio and electronics in his teens. After graduating from Polytechnik of Gdansk in Poland, he designed a number of valve and solid-state power and high power amplifiers. He moved to Australia, and in 1990 his first commercial Speaker System Designer (SSD) DOS program became available, first in Australia and then in the US and Europe. In mid '90, he replaced SSD with the now well-recognized SoundEasy. Bohdan lives in the eastern suburbs of Melbourne with his wife and three children.

Merlin Blencowe ("An Optimized SRPP Amp, Part 2," p. 18) is a young technical author specializing in valve technology. He is currently writing his second book on this topic.

James Frane (Review: Briston BCD-1 CD Player, p. 22) has written for *The Audiophile Voice* from at least 2002 to present.



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New York AES 2009

The audio industry feels the economic pinch.

The 127th Convention of the Audio Engineering Society was held in New York City over the long Columbus Day weekend, Oct. 9-12, 2009. The state of the world economy was alarmingly evident at the 2009 convention. I usually arrive at the Javits Convention Center late Thursday afternoon to beat the crowds at the registration desks, which normally occupy a large space in the lobby. When I arrived for this convention, there was no sign of AES anywhere in the lobby—the registration desks had been moved into the exhibit area, occupying

space normally reserved for the large, high-priced exhibits found at the heads of the exhibit aisles.

In recent years, there have been 11 aisles of exhibits at AES, with several demonstration rooms off the main exhibit area, and two corridors of demo rooms on the lower level of Javits. This year, there were eight shortened aisles of exhibits and no demonstration rooms. One manufacturer commented to me that no one was willing to spend the “tens of thousands” of dollars that it costs to rent a demo room for the four days. Even Sony Corporation,

which normally has a large exhibit area at the head of one of the aisles, and has often rented a demo room, shared a small booth with one of their dealers.

Only one semiconductor manufacturer exhibited at this year’s AES: THAT Corporation. Analog Devices and Cirrus Logic, regular exhibitors in the past, were absent, as were TI/Burr-Brown and National Semiconductor. Cirrus was listed in the Exhibitor Directory, but their booth was empty—they must have pulled the plug at the last minute. By my count, there were just under 320 exhibitors this year, compared to nearly 450 in 2007. The convention itself seemed to lack the usual focus—even the convention theme, “Making the Right Connections,” was rather generic and left me with the impression that they couldn’t think of anything better. Nonetheless, there were many very worthwhile exhibits, presentations, and workshops.

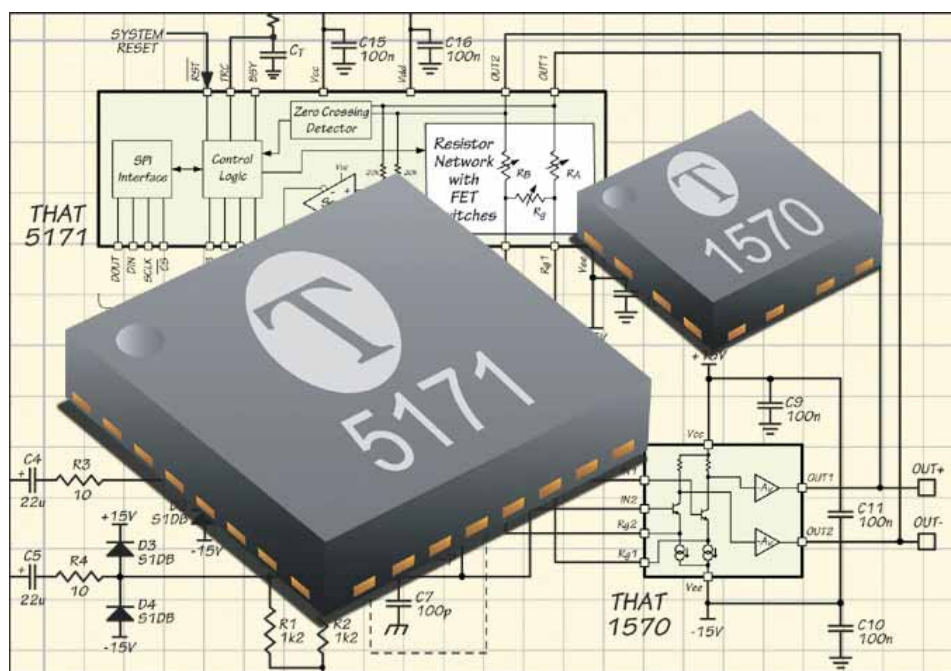


PHOTO 1: THAT 1570 Programmable Microphone Preamplifier and 1571 Digital Preamplifier Controller. These high-performance chips are manufactured in small QFN packages (Quad Flat No leads) to minimize PC board real estate. (Courtesy of THAT Corp.)

THAT SEMINAR

THAT Corporation (www.thatcorp.com) presented another excellent manufacturer's seminar, a sequel to their "Analog Secrets Your Mother Never Told You," which they featured at the 2007 and 2008 conventions. At this year's presentation, they discussed their new 1570 high-performance microphone preamplifier (**Photo 1** and **Fig. 1**). The 1570 employs a current-feedback amplifier and has a slew rate of $53\text{V}/\mu\text{s}$.

Noise is an impressive $1\text{nV}/\sqrt{\text{Hz}}$ at 60dB gain, THD+N is 0.0008%, and bandwidth is 4.2MHz. Gain is adjustable up to 60dB, with three external resistors, but the 1570 can also be mated with their 1571 Digital Preamplifier Controller IC, which allows programmable gain of +5.6dB and +13.6 to 68.6dB in 1dB steps. Four general-purpose control outputs can be used

to control a variety of peripheral functions, including muting, phantom power switches, signal routing switches, LED indicators, and so on.

The company offers this description of the 1570 on their website: "Designed from the ground up in THAT's complementary bipolar dielectric-isolation process and including laser-trimmed Si-Chrome thin-film resistors, the 1570 improves on existing integrated microphone preamps by offering more versatile gain configuration, lower noise at low gains, higher slew rate, and lower distortion." The 1570 has differential inputs and outputs and, unlike many IC-based mike preamps, it can operate on rail voltages as high as $\pm 17\text{V}$, giving it a dynamic range of 130dB.

Back in 2008, Sadie, manufacturer of one of the finest high-end digital audio editing systems, came close to

closing its doors. Fortunately, Prism Sound rescued the company and Sadie is back, alive and well (www.prismsound.com; www.sadie.com). Sadie editing systems have typically run in the five-figure range, in part due to the need for proprietary computer hardware. This year, Sadie introduced their first editing system designed to operate on any Windows-platform computer, though Sadie recommends a Prism Sound Orpheus sound card (**Photo 2**).

The Sadie 6 system is available in four configurations: Radio Producer, Post Suite, Mastering Suite, and Sound Suite. The Sound Suite is the premium version, which includes all plug-ins offered by the company. The Sadie 6 software can also be used with their own proprietary hardware, such as the PCM-H8 Digital Audio Workstation.

FIGURE 1: THAT 1570: THAT 1570 and 1571 in a typical microphone preamp configuration with phantom powering and differential outputs. (Preliminary drawing, Courtesy of THAT Corp.)

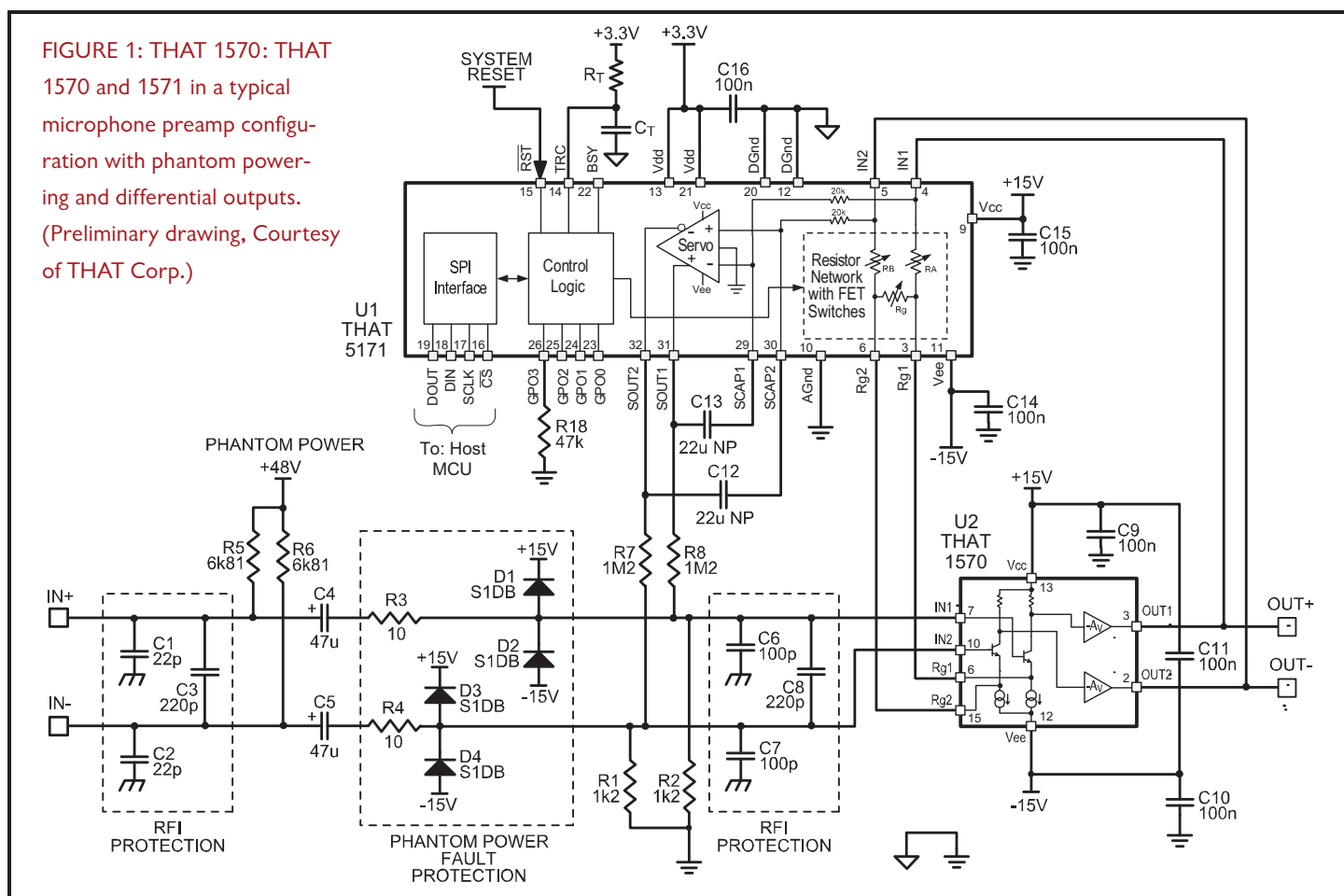




PHOTO 2: Now under the ownership of Prism Sound, Sadie has introduced Sadie 6, a digital audio recording and production system that runs on a standard Windows platform. The Sadie 6 software can also be used with their proprietary hardware, such as the PCM-H8 Digital Audio Workstation. (Courtesy of Prism Media Products Ltd.)

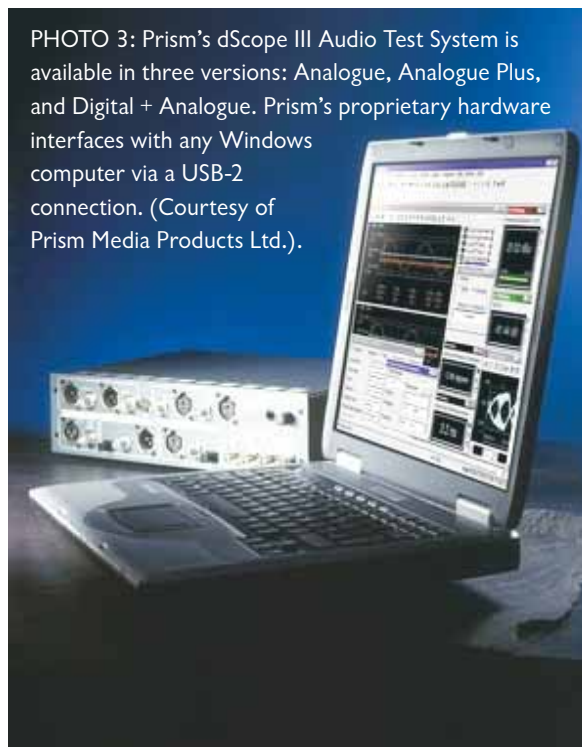
Though Prism Sound has gained a reputation for their outstanding A/D and D/A converters, they have also become a leader in the field of audio measurement. This year they were featuring their dScope III Audio Test System, including a new, low-cost version (**Photo 3**).

The test system comes in three versions: dScope Series III Digital + Analog Audio Test System, which is the original full version with extensive digital and analog capabilities; dScope Series IIIA+ Analog-Plus Audio Test System for those who don't need digital audio capabilities, but do need the advanced features of the full dScope Series III; and dScope Series IIIA Analog Audio Test System for those who don't need digital audio capabilities or the full power of dScope analysis. All standard audio measurements are supported, including amplitude, frequency, inter-channel phase, distortion, sweeps of up to four measurements simultane-

ously, up to 40 simultaneous FFT derived measurements, digital interface testing, limit checking, both scalar values and traces, real-time oscilloscope display, real-time frequency spectrum display, and filter shape display.

There is little new in audio recording these days, at least no significant

PHOTO 3: Prism's dScope III Audio Test System is available in three versions: Analogue, Analogue Plus, and Digital + Analogue. Prism's proprietary hardware interfaces with any Windows computer via a USB-2 connection. (Courtesy of Prism Media Products Ltd.).



improvements. The industry has not seen the need to move beyond 192kHz/24-bit PCM, or the DSD format used for SACDs. Digital eXtreme Definition (DXD), developed by Pyramix/Merging Technologies, is a PCM system that does 32-bit floating, 352.8kHz recording, but it's used primarily for mastering and editing of high-resolution recordings prior to conversion to DSD for release on SACDs (www.merging.com).

Even DXD has been around for at least two years, and was on exhibit at AES in 2007. Version 6.1, exhibited this year, offered workflow improvements over V. 6.0. Given the fact that the consumer has shown little interest in high-res digital, the industry really has had no incentive to take digital audio any further.

What's changing is physical size of high-performance audio recorders, and manufacturers are making high-resolution digital recording more portable than ever. Sony featured their PCM-M10 palm-sized professional digital recorder, which records linear PCM at sample rates up to 96kHz with up to 24-bit resolution (**Photo 4**).

Linear PCM data is stored as WAV files, but the PCM-M10 can also record in MP3 format, for non-critical applications. It has 4GB of built-in flash memory, which can store up to six hours of CD-quality (44.1kHz/16-bit) audio.

It also has a dual MicroSD and Memory Stick Micro (M2) slot, for additional memory options. The recorder has a pair of built-in omnidirectional condenser microphones, and a built-in speaker. A stereo mini jack allows connection of external microphones, and there's also a stereo line-level input, and a line-level output. The PCM-M10



PHOTO 4: Sony's PCM-M10 palm-sized professional digital recorder, which records linear PCM at sample rates up to 96kHz with up to 24-bit resolution. The recorder has 4GB flash memory, and a pair of built-in omnidirectional microphones. (Courtesy of Sony Corp.)

requires 3V to operate, which can be supplied by a pair of AA batteries, or the included external AC adaptor.

Tascam (www.tascam.com) was exhibiting their HS-P82 8-track digital field recorder (**Photo 5**). The unit records on dual Compact Flash cards, with no moving parts, and supports eight-track recording at up to 96kHz/24-bit, and four-track at 192kHz/24-bit. It will also record a stereo mixdown. Inputs include eight high-quality microphone preamps with +48V phantom powering.

The eight XLR input connectors can also function as line inputs, and stereo XLR line outputs are also included. Digital I/O includes eight AES/EBU digital inputs and outputs, via a DB-25 connector, and a stereo digital BNC

output. Each AES/EBU digital input is equipped with a sample rate converter, allowing compatibility with virtually any standard digital audio datastream.

The recorder also has SMPTE time code and word sync in and out, via BNC connectors. A PS-2 keyboard connector facilitates naming of tracks, and a USB-2 connector allows high-speed data transfer to computers. The recorder stores data as Broadcast WAV files that include iXML metadata, allowing quick import into nearly any video or audio editing system. Several powering options are available, including AA (X10) or NP batteries, the supplied AC adapter, external DC input, or a V-mount adapter for Endura batteries. The HS-P82 is housed in a rugged, lightweight aluminum chassis. With a list price of \$6500, it's certainly not cheap, but the HS-P82 is a breakthrough in high-performance multi-track portability.

HISTORICAL PRESENTATIONS

Among the historical presentations, Tom Fine's "Mercury Living Presence" traced the history of this record label, whose classical recordings are still highly regarded by audio-

philes for their extraordinary realism, and by music lovers for the many outstanding performances captured by Mercury. Tom is the youngest son of the late Mercury engineer C. Robert Fine and producer Wilma Cozart Fine (sadly, Wilma passed away only two weeks before this convention). Tom's presentation was filled with photographs and illustrations documenting the development of the fabled Mercury sound from their single-microphone beginnings in the late 1940s, through the three-spaced omnidirectional technique first captured on 1/2", three-track tape, and then on 3-track, 35mm magnetic film. He also discussed the CD reissues of the classic Mercury recordings for which his mother returned from retirement to perform the analog-to-digital transfers. Tom also played many recorded examples along the way.

In recent years, AES has been devoting one room to presentations requiring a good audio system, complete with PMC M2-series loudspeakers that are capable of filling a very large room with surprisingly good sound. Fortunately, Tom's presentation was held in this room, allowing the audience to



PHOTO 5: Tascam's HS-P82 8-track digital field recorder stores digital audio on dual Compact Flash cards, with no moving parts. It supports 8-track recording at up to 96kHz/24-bit, 4-track at 192kHz/24-bit, and will also record a stereo mix down. (Courtesy of Tascam).

really hear the examples he played. This was an excellent presentation on one of the most important chapters in what many of us regard as the Golden Age of classical recording.

Those who are active in the Association for Recorded Sound Collections (ARSC) are by now well familiar with the research of David Giovannoni (**Photo 6**). At the 2008 ARSC Conference, held at Stanford University in Palo Alto, Calif., Giovannoni and his colleagues at First Sounds reported on the research they've conducted on Edouard-Léon Scott de Martinville, the inventor of the Phonautograph. The Phonautograph was a recording device that etched lateral sound vibrations on paper coated with lamp black. The paper was wrapped around a drum that

was hand-turned. The inventor had no means of playing back these Phonautograms, as they're called—the purpose was to provide a means of visual analysis of sound vibrations, a precursor to the modern-day oscilloscope.

Giovannoni and his colleagues, during the course of research at French Academy of Sciences in Paris, discovered that many of Scott's Phonautograms still existed and, by scanning and digitizing them, they could be played for the first time! Several digitally restored phonautograms were played at the ARSC conferences in 2008, the earliest dating from 1860, 17 years prior to Edison's invention of the phonograph. Giovannoni's AES presentation "Before Edison" offered attendees an update on their ongoing

research, including the discovery of even earlier recordings. Anyone interested in sound recording history should visit their website www.first-sounds.org, where you can hear sound clips of some of these recordings.

Three faculty members from the University of Tokyo—Teruo Muraoka, Takahiro Miura and Tohru Ifukube—contributed to a poster session titled "Frequency Characteristics Measurements of 78-rpm Acoustic Record Players by the Pulse Train Method" (Mr. Muraoka was one of the developers of the CD-4 Quadraphonic LP record; his Pulse Train measurement method dates back to 1973, and was originally used for phase measurements on the disc recording and reproduction system). The authors



PHOTO 6: David Giovannoni examines one of Léon Scott de Martinville's 1860 phonautograms in the archives of the *Académie des Sciences* of the *Institut de France*, where it was deposited in 1861 (left); detail of an 1859 phonautogram made by Scott and included in his patent paperwork, preserved at the *Institut National de la Propriété Industrielle* (INPI), the French patent office (right). (Courtesy of First Sounds, www.firstsounds.org; photos by Isabelle Trocheris)

have been involved in research on restoration of seriously damaged audio signals using a technique known as Generalized Harmonic Analysis. The premise of the authors is that, in order to properly reproduce vintage recordings, it is important to know the frequency response characteristics of contemporary reproducing equipment. The flaw in this argument is that phonographs manufactured during the 78-rpm era, especially during the era of acoustical recording (up to c. 1925), were not capable of revealing all that was contained in the record grooves, and often the frequency response of an old phonograph has little to do with the most optimum playback characteristics for vintage records.

The authors claim that the Pulse Train method gives a more precise measurement of the frequency response characteristics of whatever is being measured, whether a loudspeaker or an acoustic phonograph. They made their own Pulse Train record by putting eight radial scratches on a 78-rpm shellac record. That record was used to measure the frequency response of a Victor “Credenza,” an acoustic phonograph invented by Maxfield and Harrison at Bell Labs for the reproduction of early electrical recordings (the “Credenza” was the first phonograph with an exponential horn). The authors provide a frequency response graph for the Credenza that they claim is more accurate than the original graph published by Maxfield and Harrison in 1926.

There are many English language difficulties in their convention pre-print, as well as in the handout they distributed, so I hope my summary is reasonably accurate. It is difficult to determine what they consider to be an optimum restoration of an old recording. Whether they are trying to reproduce the sound of an old phono-

graph or compensate for its deficiencies is unclear.

Mr. Muraoka generously gave me a CD he produced titled “Super Nikisch: Monumental Recordings 1913-1920.” The disc contains recordings by the legendary conductor Arthur Nikisch, recordings with which I am very familiar from various reissues, as well as a vinyl 78-rpm pressing made from original metal parts of his 1914 *Oberon Overture* by Weber. The sonic results on the CD I received were very disturbing. Throughout, the music is accompanied by serious artifacts from the digital signal processing, including swirling sounds, comb filtering, plus strange-sounding pops and dropouts that are probably by-products of the overzealous removal of clicks and pops on the original records.

ELECTRONICS FOR RECORDING STUDENTS

Michael Stucker, a faculty member at Indiana University, chaired the workshop “Teaching Electronics to Recording Students—Why Bother?” Panelists included Brian Bender of Motherbrain (substituting for Eric Brengle of Swinghouse Studios, who was unable to attend), Eddie Ciletti from Tangible Technology, Dale Manquen from Manco, Walter Sear from Sear Sound, and Jensen Transformers’ President Bill Whitlock. The panel focused on integrating electronics into an audio recording curriculum.

Several “basics” were deemed essential areas of study, including voltage, current (conventional and electron), power, AC and DC, series and parallel circuits, passive components (resistors, capacitors, inductors, transformers), active components (tubes, bipolar and FET transistors, op amps and other ICs), reading schematics, and safety grounds. More advanced theory would follow, with

math background being important. Graphing skills are also important, for frequency response, transfer functions, and so on.

Many of the panelists emphasized the need for hands-on experience with electronics, which few young people have these days. Whitlock noted that this is not a plug-and-play world. There are many badly-designed products on the market, and students need to be educated in proper troubleshooting techniques when dealing with interfacing problems, including those caused by a frequent source of hum: the “pin 1” problem. In identifying prospective students for a recording engineering program, latent interests are important. Students should be interested in working with their hands; they should be inquisitive about technology, and interested in taking old equipment apart. Altogether, this was a very worthwhile session.

BLU-RAY AUDIO

Stefan Bock, of msm-studios in Germany, chaired the session “Blu-ray as a High Resolution Format for Stereo and Surround.” Panelists included Gary Epstein from Dolby Labs, Tom McAndrew from DTS, Johannes Müller from msm-studios, and Ronald Prent from Galaxy Studios in Belgium. The company represented by Bock and Müller, msm-studios, is promoting the Pure Audio Blu-ray format for high-resolution stereo and surround audio. The Pure Audio format was designed to make Blu-ray audio playback as simple as a CD. Unlike DVD-Audio, which normally requires that you operate your television in order to play an audio-only disc, Pure Audio Blu-ray is an autonomous audio medium. Colored keys on the remote control select the desired program, whether stereo or surround, DTS, Dolby Digital or linear PCM.

During his presentation, Müller noted that SACD remains a niche market, requiring special players, and the audio market simply isn't big enough to have its own HD format. Part of Blu-ray's appeal is that it isn't just an audio format—it's also the high-end format for video, and offers a chance to finally produce a high-def audio standard for stereo and surround. As a demonstration, he played a recording of the Grieg Piano Concerto, with Percy Grainger's Duo-Art reproducing piano roll made in the early 1920s accompanied by the Kristiansland Symphony Orchestra conducted by Rolf Gupta. Several audio discs, including the Grieg, are currently available on the Pure Audio website (www.pureaudio-bluray.com).

Twice during this panel it was stated that DVD-Audio has a maximum sampling rate of 96kHz in stereo, which is not true. During Q&A I pointed out that DVD-Audio supports 192kHz sampling rates in stereo—the 96kHz limit is true only for surround (the only difference between Blu-ray and DVD-Audio is that Blu-ray allows 192kHz/24-bit in surround). For many audiophiles, the lack of consumer interest in high-resolution digital formats such as DVD-Audio and SACD is frustrating. Epstein acknowledged the difficulty of promoting Blu-ray as an audio-only format when DVD-Audio didn't make it. Bock and Müller really didn't have an answer to the question of overcoming consumer ambivalence toward high-quality sound.

Epstein and McAndrew each offered their thoughts on why the HD audio formats developed by their respective companies are superior to the competition. DTS is against linear PCM on Blu-ray discs, noting that DTS HD is a lossless CODEC (www.dts.com). The same is true of Dolby True HD, of course (www.dolby.com).

Prent expressed his preference for linear PCM, noting its superior sync. He played an excerpt from *Black Symphony*, featuring the Dutch band Within Temptation and the Metropole Orchestra. Everyone in the orchestra was miked, producing a decidedly film-like sound through the PMC M2-series loudspeakers. The demonstration switched between various audio formats—DTS, Dolby Digital, and linear PCM. Prent stated that linear PCM allows greater involvement with the concert.

GREAT PICTURE— MEDIocre SOUND

Michael Fremer, Senior Contributing Editor of *Stereophile* magazine, chaired a workshop titled "We Got the Picture. What Happened to the Sound?" Panelists included John Atkinson, long-time Editor of *Stereophile*, Steve Berkowitz of Sony/BMG, Greg Calbi of Sterling Sound, Alan Douches of West West Side Music, Bob Ludwig of Gateway Mastering & DVD, EvaAnna Manley of Manley Audio Labs, and Randy Ezratty. The panel focused on the irony in today's audio and video preferences. Consumers are enamored, and rightly so, of 1080p high-definition pictures, yet are also satisfied with audio resolutions lower than that of conventional CDs. The general public finds compressed digital standards, such as MP3, more than adequate for music listening.

Ludwig stressed the convenience factor—the ease of MP3 playback versus CDs, SACDs, and DVD-Audio discs is comparable to the cassette tape versus the LP years ago. Ludwig also pointed out that SACD won the high-res audio format war, with 360 releases in that format last year compared to 15 on DVD-Audio. I didn't check the figures, but at 360 releases SACD may have won the war, but it's nonetheless a fringe audiophile format, and DVD-

Audio seems all but extinct. Mastering is now geared toward earbuds or earphones, and not high-quality loudspeakers. Consumers want free music and they're getting what they pay for.

Ludwig also recalled the old AR demonstration room in Grand Central Station, asking why we can't have something similar today. Fremer said that he often invites friends of his own kids into his listening room, to hear their favorite music on a high-resolution system from uncompressed sources. Their reactions have been one of amazement—"I had no idea music could sound like this!," or words to that effect. Perhaps an AR-style listening room in a prominent public location would help educate today's listeners and excite them about audio as much as they have become excited about video. Someone would need to foot the bill, of course. Fremer also pointed out that very few people listen to music to the exclusion of all else.

In my 2007 AES convention report, I discussed an article by two members of the Boston Audio Society—E. Brad Meyer and David R. Moran—who claim that double-blind, ABX testing fails to reveal any differences between CDs and high-resolution digital formats^{1,2}. Their conclusion was that there are no audible differences. In that AES report, I concluded that either the participants can't hear, or there's something wrong with the test. Atkinson noted that ABX testing fails to reveal differences between high-res and 16-bit/44.1kHz digital, yet conventional listening consistently reveals those differences. One audience member expressed the view that double-blind testing changes perception vis-à-vis conventional listening.

Bravo to Berkowitz for stating that, yes, there are audible differences between digital audio at 44.1kHz/16-bit, and 96kHz, 24-bit. Ludwig mentioned

the audible degradation introduced by jitter when a glass master is made faster than real-time. In his view, even 4× is sonically unacceptable. It was an altogether lively and worthwhile discussion.

AUDIO MYTHS

There's always been a strong contingent in AES who are skeptical, and in some cases downright hostile, to the idea that there are audible differences between amplifiers, cables, and high- versus low-resolution digital formats, and so on. That skepticism was reflected in the workshop "Audio Myths—Defining What Affects Audio Reproduction," chaired by Ethan Winer of RealTraps, with the panelists Jason Bradley of Intel Corporation, Poppy Crum of The Johns Hopkins University School of Medicine, and James Johnston of DTS.

Crum, a neuroscientist, discussed how the ear processes sound. Dr. Crum described the Haas Effect, the McGurk Effect, and auditory continuity (how the ear/brain fill in missing information, which is why music sounds continuous even when a record has clicks and pops). Crum is an acknowledged authority in her field, and while no rational attendee would take issue with

anything she said, there was a definite disconnect between her presentation and basic premise of this session. I believe the other panelists were hoping that her portion of the presentation would lend credence to some of their own agenda items, but it was not to be, at least not from where I sat.

The panel defended double-blind testing, and Winer snickered at a variety of audiophile tweeks, including Brilliant Pebbles (www.machinadynamica.com), Acoustic Art resonators (www.synergisticresearch.com), high-end power cords, and LP and CD demagnetizers, some of which I also find hard to swallow. (Full disclosure: Acoustic Art resonators are designed for treating room acoustics, and Winer's company RealTraps [www.realtraps.com] also manufactures room treatment products.)

Winer's argument that there is much more distortion in loudspeakers than in A/D and D/A converters just doesn't hold water, because the types of distortions produced by each are very different, and the distortion in a good loudspeaker will not necessarily mask the distortions produced by digital converters.

Winer described four audio param-

eters:

- Frequency response
- Distortion, including THD, IM, and aliasing "birdies"
- Noise, including hiss, hum, buzz, and vinyl crackles
- Time-based errors, including wow, flutter, and jitter

He also stated that transparency is not the only goal, and that sometimes euphonic coloration is desirable. The latter is certainly true, as evidenced by the popularity of tube, zero-feedback and single-ended designs in both high-end consumer and pro audio gear. Winer also spent some time on artifact audibility, but the audio demos that he played were virtually useless in that room on the decidedly low-end audio system. WAV files used for those demos, along with an explanation, are available on his website (www.ethanwiner.com/audibility.html). Winer disagrees with Bob Ludwig, stating that jitter is inaudible, as is dither (though he still recommends that you use it!).

However you may view the never-ending debate on what is and isn't audible, one thing is fairly certain: AES conventions are much more balanced than they were, say, 25 years ago. The workshops and presentations offered at AES conventions over the past 20 years have increasingly reflected the diversity of viewpoints prevalent at AES. In my view, this has only made the organization stronger. *ax*

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THAT's Balanced Line Drivers and Receivers

Getting the most from these high-performance chips.

By Gary Galo

During the past year, I completed the design and construction of an archival phono preamplifier, which provides variable equalization along with several other operational necessities for playing vintage records (78-rpm discs, pre-RIAA LPs, lacquer transcription discs, and so on). When it came to selecting topologies for the balanced inputs and outputs, I ruled out transformers due to high cost, physical bulk, and weight. The most practical choice, in terms of size and cost, are the balanced line driver and receiver ICs made by several IC manufacturers.

After surveying the field, it became clear that the products from THAT Corporation offer superior performance to any of their competitors' chips. Like IC op amps, these devices are capable of varying degrees of performance depending on the quality of the passive components used in the associated circuitry, and power supplies. This article describes how to get the most from these chips. Anyone intending to use them should download the datasheets for these devices on THAT's website (www.thatcorp.com).

During the October 2005 convention of the Audio Engineering Society (AES), I attended a seminar given by Bill Whitlock, President of Jensen Transformers, on "Grounding and

System Interfacing." I gave a report on this presentation in the January 2007 issue of *aX*.¹ Near the end of his presentation, Whitlock described the InGenius series of balanced line receiver ICs he designed for THAT Corporation. Whitlock developed a unique bootstrapping technique that raises the common-mode input impedance into the megohm region, while maintaining reasonable values for the input bias return resistors.

Raising the values of these resistors will degrade noise performance, but Whitlock's clever topology maintains low noise, keeping the resistor values low but the common-mode input impedance very high. The three chips in the InGenius line are the THAT 1200, 1203, and 1206, which are identical except for gain: 0dB for the 1200, -3dB for the 1203, and -6dB for the 1206.

THAT Corporation gave a seminar at the October 2007 AES convention covering—among other things—their OutSmarts line of balanced line drivers. I reported on this seminar in the June 2008 issue of *aX*.² The OutSmarts line, which includes THAT's 1606 and 1646 devices, incorporates a dual feedback loop design which prevents excessive ground currents typical of cross-coupled output stages when clipping into single-ended loads. In fact,

when driving single-ended loads, the OutSmarts devices yield their lowest noise and distortion when one output leg is grounded.

THAT 1200 OFFSET

The maximum output offset voltage for the THAT 1200-series chips is specified as 10mV, and most samples I've measured tend to run between 4 and 7mV. This may be fine for some applications, but, in the two locations I use them in my archival phono preamplifier, they are followed by DC-coupled gain stages. The typical "pro" audio solution to DC offsets is to fill equipment with electrolytic coupling capacitors, which is simply unacceptable in a piece of high-end audio equipment.

I e-mailed Gary Hebert, Chief Technology Officer at THAT, and asked whether it was possible to null the 1200's offset with a DC-servo integrator. He said that it should be possible to do so, applying the servo output to the Reference pin (pin 1). He noted one caveat: If you follow the integrator with the usual resistive attenuator, you'll seriously degrade the common-mode rejection of the chip. As you'll see shortly, the 1200-series chips actually contain the resistive divider internally, but the resistor values are such that the integrator must have a

considerably longer time constant in order to achieve the same -3dB point for the circuit as it would in a more conventional implementation.

Integrators configured as DC servos for audio amplifiers have received scant coverage in electronics textbooks and periodicals. The best tutorial on their design was published by Walt Jung in Analog Devices' massive 1993 *Systems Applications Guide*.³

Figure 1 shows an op amp configured as a unity-gain inverting amplifier with DC-servo control of the circuit.

The unity-gain inverter consists of IC1, R1, and R2 (Gain = $R2/R1$). IC2 is configured as an integrator, with its time constant determined by R5 and C1 ($T = RC$). In this application, the integrator functions as a DC-servo amplifier, applying DC feedback to the non-inverting input of IC1 through the resistive divider (attenuator) R3 and R4.

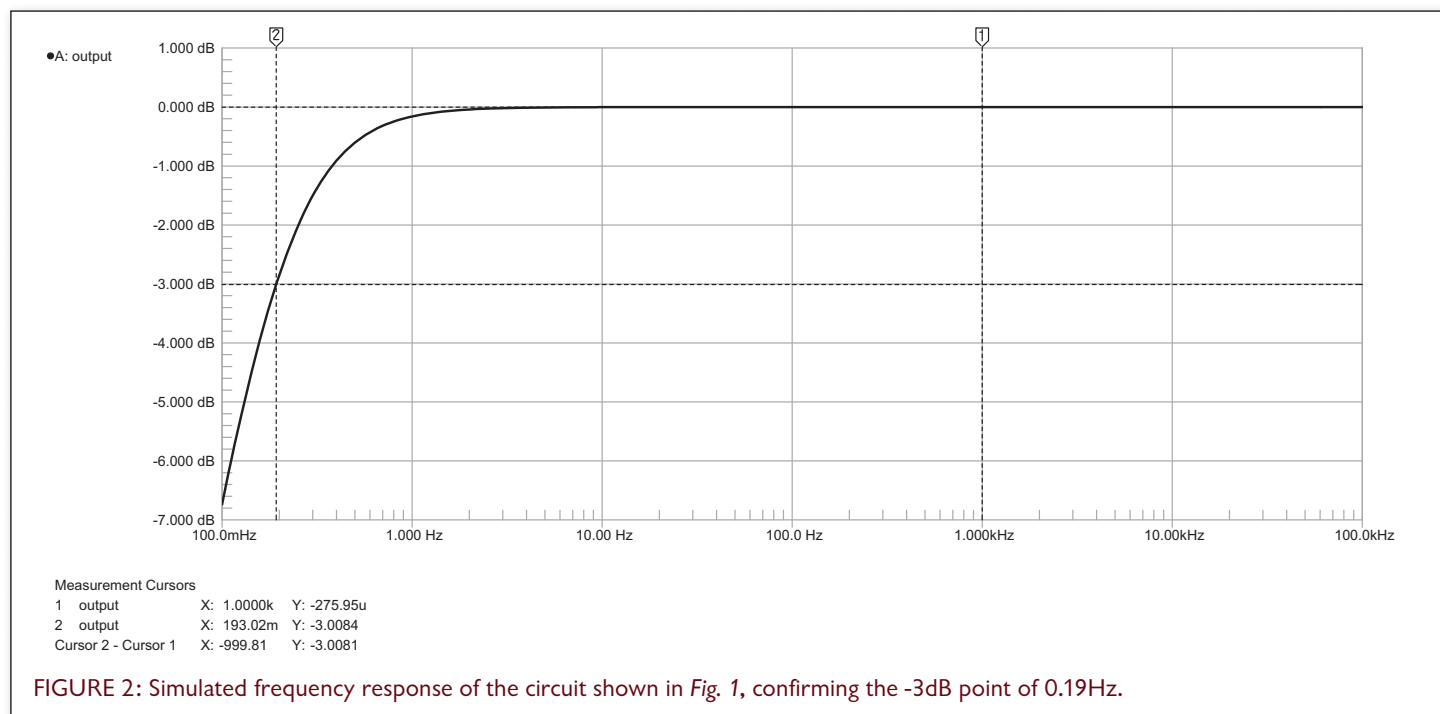
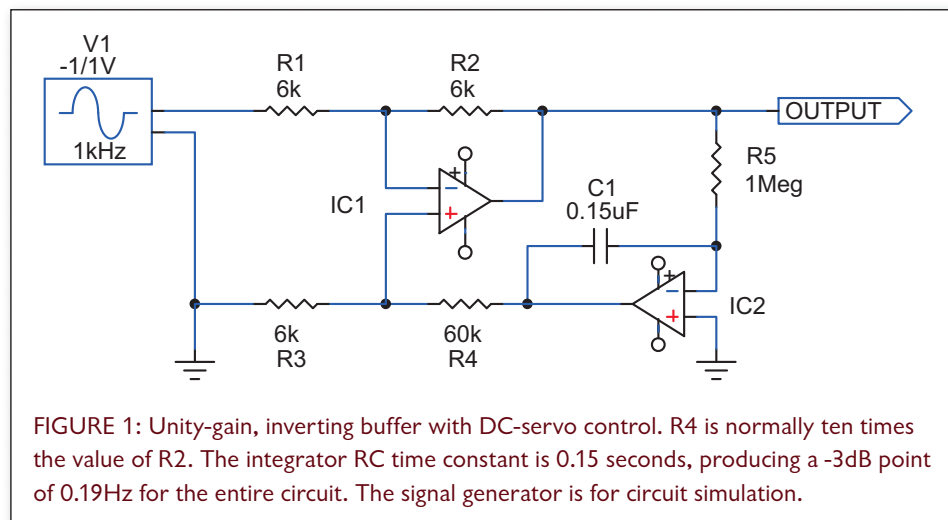
Following Jung's guidelines, R3 is equal to R1, and R4 is ten times the value of R2. The -3dB point for the entire circuit should be about 1/100

of the lowest audible frequency—usually 20Hz—in order to minimize rolloff and phase shift within the audible spectrum. R5 and C1 are selected for reasonable values; R5 will normally be at least 500k and the R5/C1 combination should yield a -3dB point for the entire circuit of around 0.2Hz.

Following the math outlined in Jung's article, which I won't repeat here, the integrator time constant for **Fig. 1** is 0.15 seconds and the -3dB point for the entire circuit is 0.19Hz. The simulated frequency response for this circuit is shown in **Fig. 2**, which verifies that the calculations are correct.

CIRCUIT SIMULATION

Figure 3 shows a simulation of the THAT 1200 balanced line receiver in a complete working circuit, including RFI protection at the input, as recommended in the manufacturer's data-sheet, which also shows methods for ESD (electrostatic discharge) protection, if needed. Internally, the THAT 1200-series chips have diodes that



provide “modest protection against common low-voltage ESD incidents.” The manufacturer’s own testing has shown that repeated exposure to levels of ESD above 1kV through the non-inverting and/or inverting inputs can degrade the common-mode rejection and, ultimately, cause failure of the chip. If your application may expose the chip to high levels of ESD, I recommend using the protection. Note that the datasheet shows one method for the 1200 and another for the 1203 and 1206, because the 1203 and 1206 allow input signals that swing higher than the supply rails.

The circuit inside the dashed line is the THAT 1200 equivalent circuit, which is shown on the first page of the datasheet. The type of op amp used in the simulation isn’t critical if all you need to do is verify that the circuit works with the servo connected, and check the low-frequency -3dB point. The generic “Op-Amp 5” model found in most Spice-based simulation programs will work fine.

When I asked Gary Hebert what

op amp would be suitable for simulating the THAT 1200, he suggested a 5532. Bear in mind, however, that the performance of the THAT 1200 is an order of magnitude better than the dated 5532, with higher slew rate (12V/μS versus 9V/μS for the TI NE5532A), and wider bandwidth (22MHz versus 10MHz). Both chips will operate with rail voltages higher than the standard ±18V. The absolute maximum rating for the NE5532A is ±22V, compared to ±20V for the THAT 1200. So, the NE5532A (and the single NE5534A) will allow you to run your simulation at the full supply rail capability of the THAT 1200. A safety margin is always desirable for long-term reliability; the ±20V maximum rating ensures that long-term operation at ±18V will be no problem.

Note the output stage consisting of OA3 plus the resistors R101 through R104. R101 and R102 are the functional equivalents of R1 and R2 in **Fig. 1**. Normally, the reference pin (1) on the THAT 1200 is grounded, so OA3

functions as a unity-gain differential amplifier with a single-ended output.

With a DC-servo integrator connected as shown in **Fig. 3**, the reference pin is still effectively at signal ground. R103 and R104 become the functional equivalents of R3 and R4 in **Fig. 1**. But, R104 is internally set at 6k, the same value as R102. If these two resistors aren’t the same value, the common-mode rejection of the chip will be severely compromised. So, R104 won’t be ten times the value of R102.

In order to make the -3dB point for the entire circuit similar to **Fig. 1**, the integrator time constant will need to be considerably longer. C5 and R4, 1μF and 750k, yield an integrator time constant of 0.75sec, which produces a -3dB point for the entire circuit of 0.212Hz. The simulated frequency response of this circuit is shown in **Fig. 4**, again verifying that the calculations are correct.

The THAT 1203 and 1206, with gains of -3dB and -6dB, respectively, may be desirable in applications where

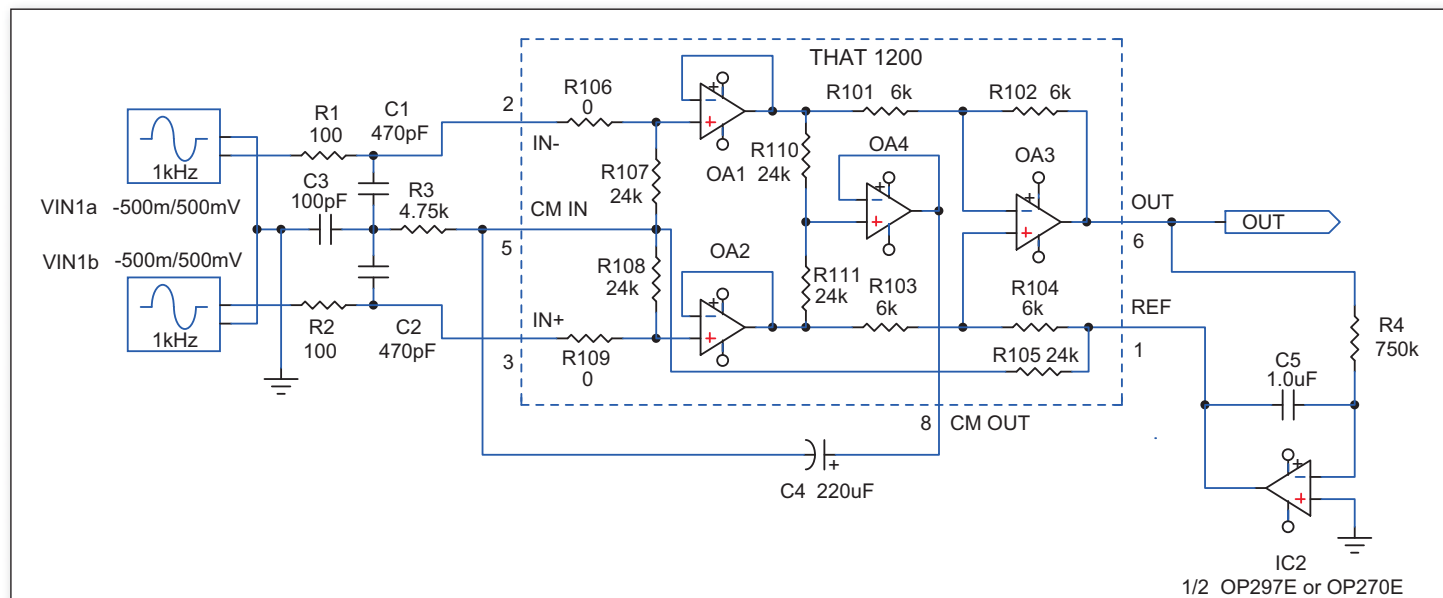


FIGURE 3: Simulation of the THAT 1200 balanced line receiver with input RFI protection and DC-servo control. R104 is internally set to the same value as R102. In order to achieve the -3dB point of 0.21Hz for the entire circuit, the integrator time constant must be set at 0.75 seconds.

higher input voltages are likely. In the 1203, R101 through R104 are the same values as in the 1200. But, in the 1206, R101 and R103 are 7k, with R102 and R104 set at 5k. This is no problem for the servo, since the math is a wash. The -3dB point for the circuit using a 1206 will be the same as it is with the 1203 and 1200.

So, you can swap 1200-series chips without having to re-calculate the integrator R/C values. In the 1203 and 1206, R106 and R109 are set at 7k, and R107 and R108 are 17k. These will obviously have no effect on the DC servo, but they do affect the overall gain of the device.

SERVO AMP CHOICES

The choice of op amp for the servo amplifier will depend on what's most important in your application—the lowest possible noise or the lowest possible output offset. Noise of the THAT 1200 is specified at an impressive -107dBu, so any servo amplifier has the potential of degrading its excellent noise performance. Gary Hebert

suggested an OP27.

For a stereo pair, I prefer to use a dual op amp to conserve space, so I chose Analog Devices' OP270E. As noted in the datasheet, the OP270E offers comparable performance to the OP27E. With the OP270, I measured noise at -96dB relative to 2V RMS out.

Input offset voltage of the OP270E is extremely low—typically 10 μ V—but input bias current is 5nA. So, the output offset of the circuit will be somewhat affected by the source impedance, set by R4, which is 750k. Output offset of the THAT 1200 using the OP270E usually runs between 1 and 2mV, which may be sufficiently low for many applications. It wasn't for one of mine, however, so I decided to try the OP297E, which I have used in many other DC servos.

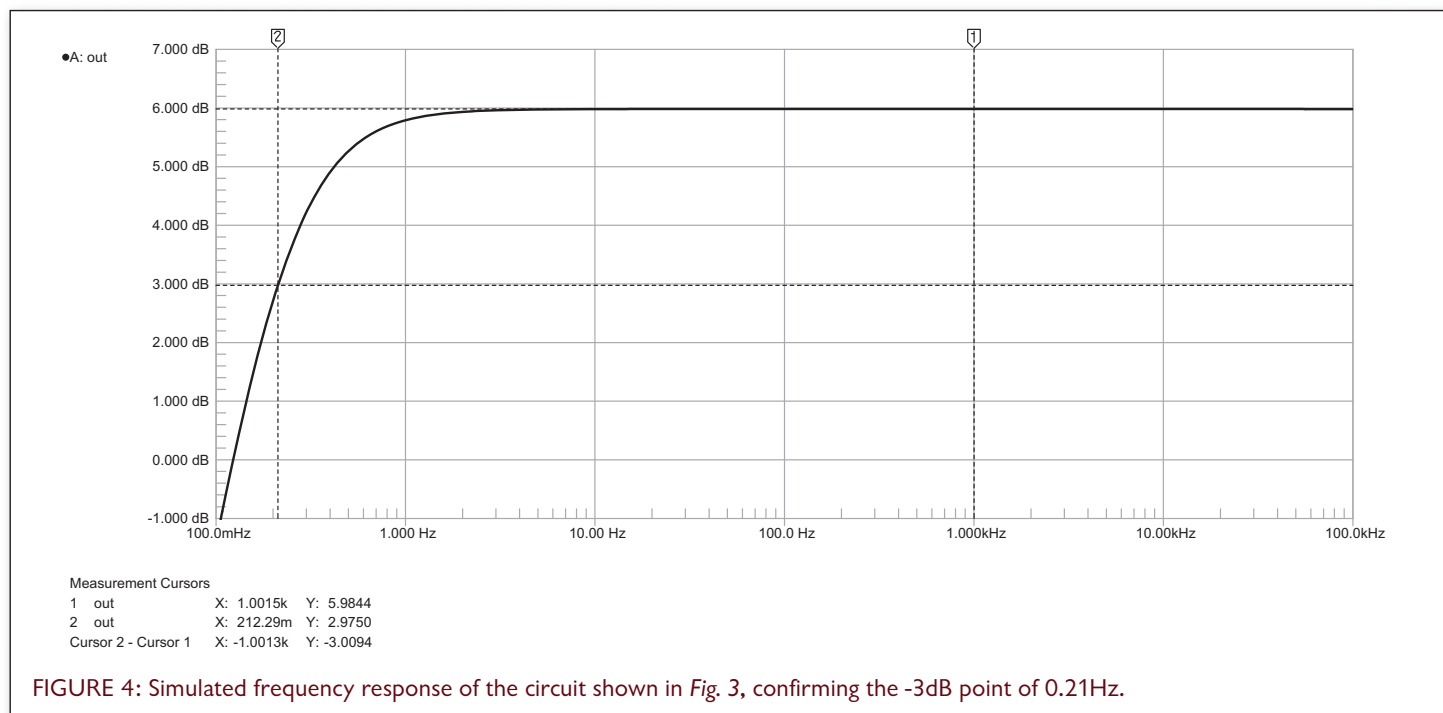
The OP297E has higher input offset voltage than that OP270E—25 μ V—but it also has ultra-low input bias current, typically 20pA. Output offset from the THAT 1200 using the OP297E is usually less than 0.2mV. Although the

OP297E is noisier than the OP270E, noise performance of the THAT 1200 is degraded no more than 2dB compared to the OP270E. For applications where ultra-low output offset is necessary, the OP297E is the best choice. Digi-Key carries only the ceramic DIP packages for the OP297E and OP270E ("Z" suffix); the plastic DIP version ("P" suffix) are certainly fine, if you can find them.

THAT 1200 CONSTRUCTION

In choosing parts for these circuits, I avoided components containing steel, because I believe that non-permeable materials have a positive effect on the sound. **Figure 5** shows a single-channel THAT 1200 balanced line driver circuit with DC-servo control, input XLR connector, and power supply bypassing. The DH Labs D4 prototyping board sold by Old Colony is ideal for building a stereo pair using a dual op amp for the servo amplifier.

Photos 1 and **2** show a recommended layout for the circuits; the



temporary leads are attached for testing. Mount the supply bypass capacitors on either end of the board. You'll need to cut traces under these capacitors to avoid shorting them out (**Photo 2**), and install jumpers to make connections to the supply rails and the center ground bus. I recommend Nichicon FG-series electrolytics bypassed with Wima or Roderstein polyesters.

I used Mil-Max machined-pin, gold-plated IC sockets for the THAT 1200 chips and the servo op amp, because I was still experimenting after my boards were built. Consider them optional. If you wish to experiment with different servo op amps, or the three 1200-series chips, I suggest soldering the ICs to an Aries gold-plated header. This gives a gold-on-gold connection when the chips are mated with the IC sockets.

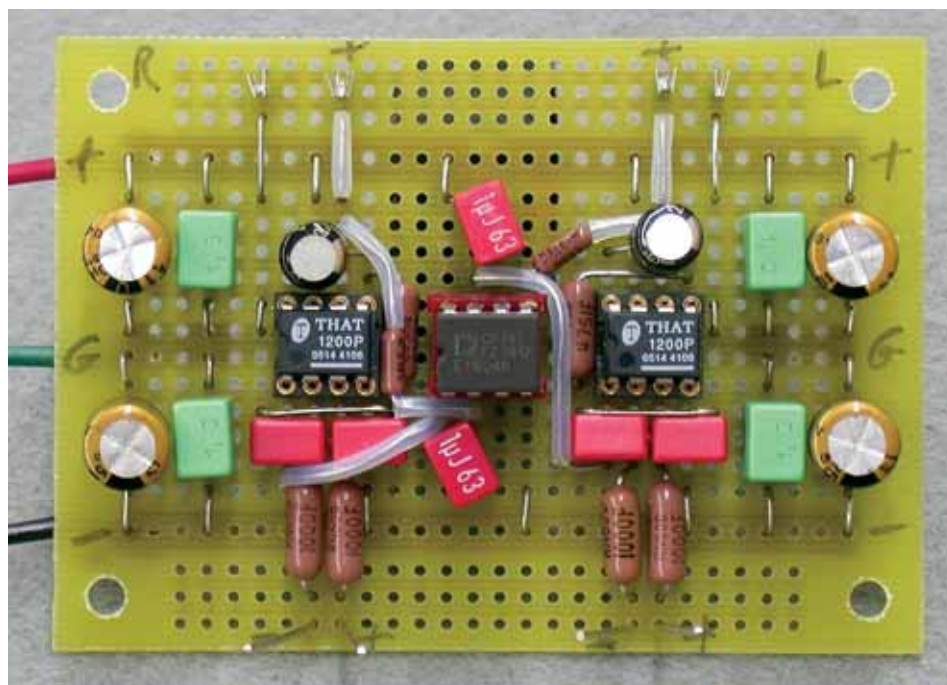


PHOTO 1: A stereo pair of THAT 1200 balanced line receivers with DC-servo control, built on DH Labs' D-4 Prototyping Board. The servo amplifier is a dual op amp mounted in the center of the board. Power supply bypass capacitors are located on each end of the board. Leads have been attached for testing.

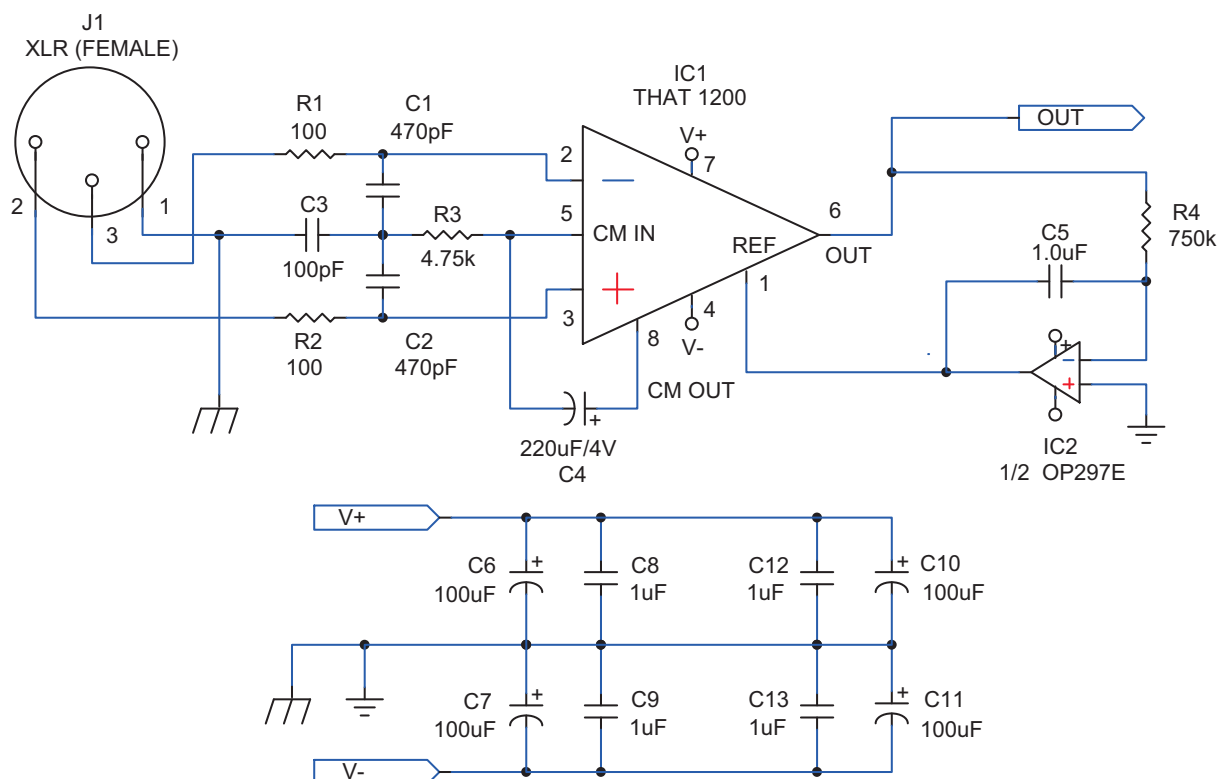


FIGURE 5: Complete circuit for the THAT 1200 balanced line receiver, including the input XLR connector and power supply bypassing. One channel is shown. Supply bypassing is sufficient for two channels.

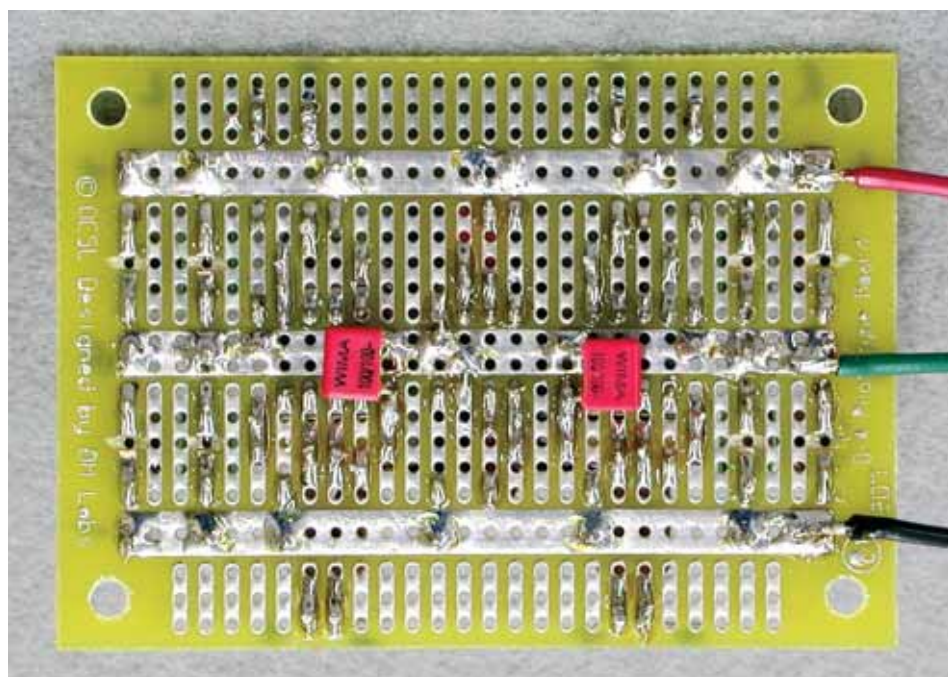


PHOTO 2: Bottom view of the THAT 1200 balanced line receiver board. The 100pF capacitors in the RFI filter, C3, are mounted on the bottom of the board.

Capacitors C1 through C3 are Wima FKP2-series polypropylenes. C3 is the only part that won't fit on the component side of the board. Mount it on the foil side, as shown in **Photo 2**. THAT Corp. specifies a 4V capacitor for bootstrap capacitor C4. The best type I've found is a Black Gate PK-series sold by Michael Percy Audio. Unfortunately, Rubycon no longer manufactures the Black Gate line, but Percy still has them in stock. When these are no longer available, I suggest substituting a Nichicon KZ-series. The smallest available is 25V, but it should still fit. Servo capacitor C5 is the same Wima or Roderstein polyester type used for power supply bypassing.

All resistors are Vishay/Dale RN60 metal film types except for R4, which

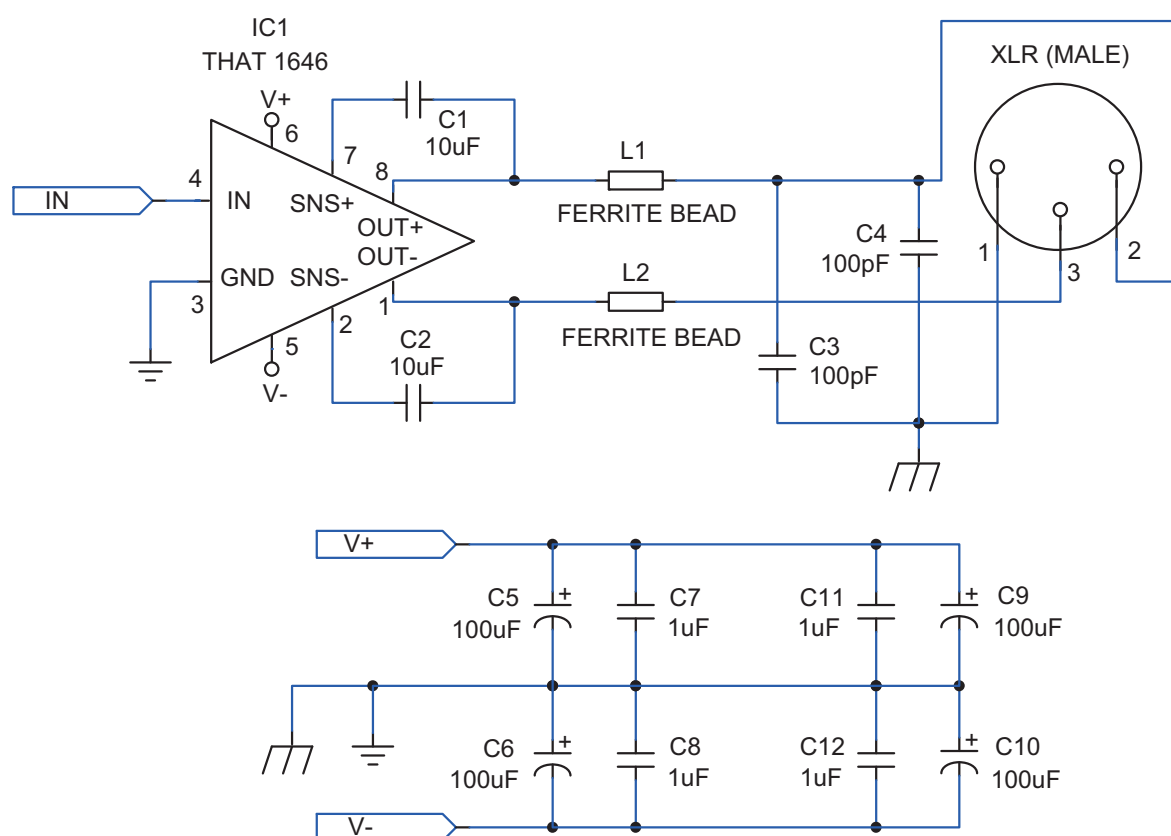


FIGURE 6: Complete circuit for the THAT 1646 balanced line driver, including the output XLR connector and power supply bypassing. One channel is shown. Supply bypassing is sufficient for two channels.

can be an RN55. I consider these to be good entry-level high-end audio resistors, particularly since all of the RN60 types still seem to be made with copper end caps (copper leads are specified, but the datasheet makes no mention of the end caps). Some of the RN55 types have been coming through with steel end caps, but the current through the 750k resistor is so low that the magnetic effects are of little consequence. Use sleeving over any leads that may touch others.

If your application demands more exotic resistors, by all means use them. The PR9372-series from Precision Resistive Products are designed specifically for audio applications and have gained an excellent reputation in high-end audio circles (www.prpinc.com/pdf/Audio_PR9372_Series.pdf). Parts Connexion carries a large assortment of these resistors at very reasonable prices. I prefer Neutrik XLR connectors with gold-plated contacts.

BALANCED LINE DRIVER

THAT's 1646 Balanced Line Driver is a very simple IC to implement, requiring very few external components. A complete schematic, including RFI protection and power supply bypassing, is shown in **Fig. 6**. A stereo pair will fit on the D4 prototyping board with room to spare; IC sockets are optional. Use the same types of capacitors for supply bypassing that I recommended for the 1200. Locate them on either end of the board, cutting the PC traces under the capacitors, and installing jumpers to make connections to the supply rails and ground bus (**Photos 3** and **4**, which also show leads attached for testing).

The common-mode offset protection capacitors C1 and C2 ensure very low output common-mode voltage

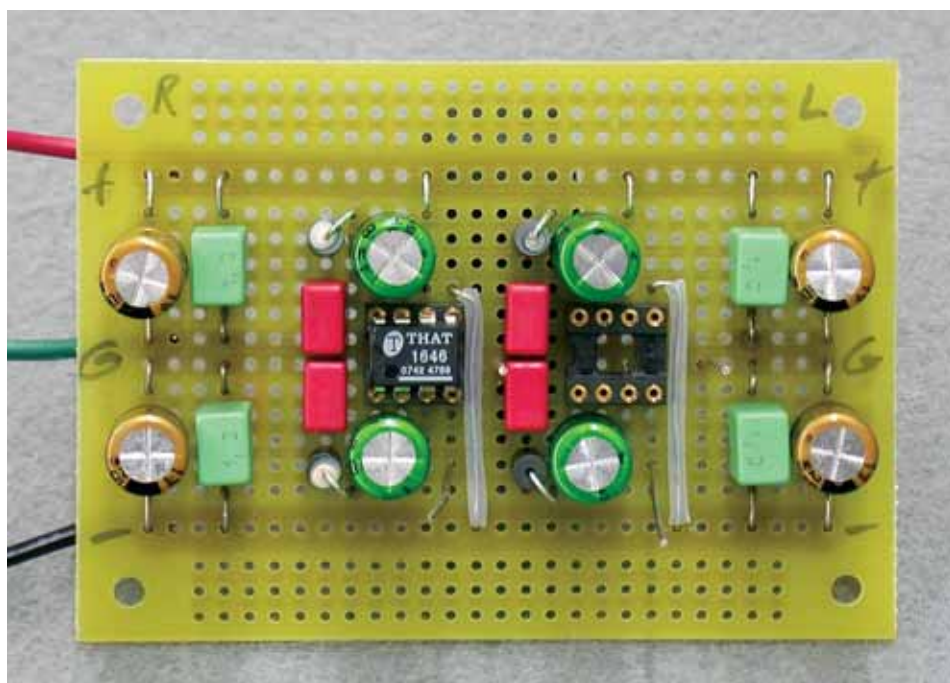


PHOTO 3: A stereo pair of THAT 1646 balanced line drivers built on DH Labs' D-4 prototyping board. For illustration, the 1646 chip has been removed from the machined-pin, gold-plated IC socket on the right. Power supply bypass capacitors are located on each end of the board. Leads have been attached for testing.

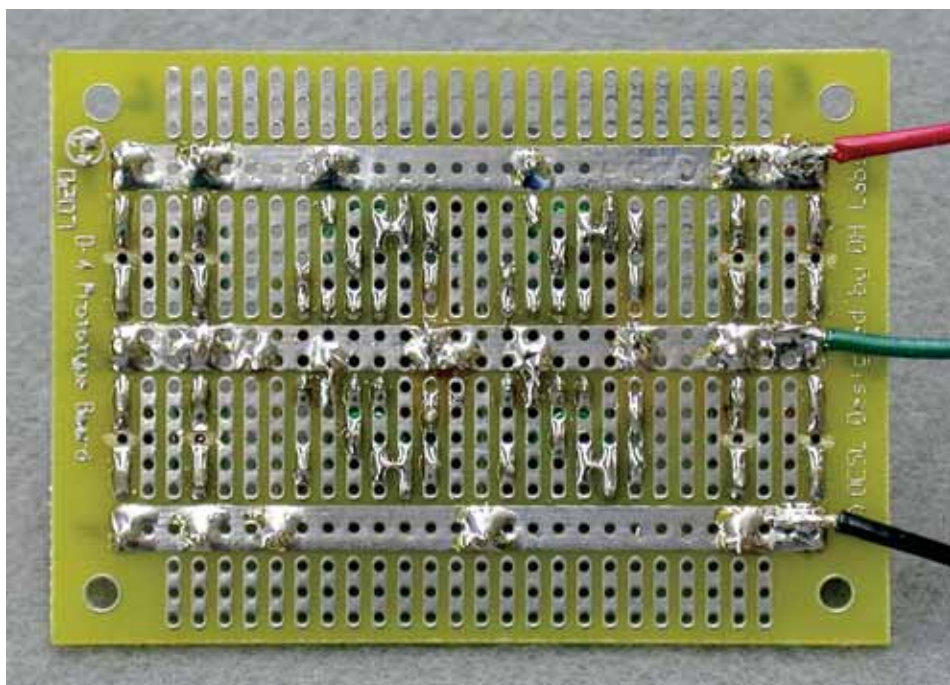


PHOTO 4: Bottom view of the THAT 1646 balanced line driver board. All parts fit on the component side of the board.

offset—typically $\pm 3.5\text{mV}$ (maximum $\pm 15\text{mV}$, though most samples I've used were within the typical range). With-

out these capacitors, typical offset will be $\pm 50\text{mV}$, and can be as high as $\pm 250\text{mV}$; even $\pm 50\text{mV}$ is unaccept

PARTS LISTS

THAT 1200 Line Receiver Board

(Quantities are for a 2-channel stereo board)

- (1) DH Labs D4 Prototype Board (Old Colony Sound Lab #PCBD-4)
- (2) THAT 1200, 1203, or 1206 Balanced Line Receiver IC (IC1; Mouser #887-1200P08-U, 887-1203P08-U or 887-1206P08-U; see text)
- (1) Analog Devices OP297E Dual Op Amp (IC2; Digi-Key OP297EZ-ND) – or – OP270EZ (Digi-Key OP270EZ-ND)
- (3) Mill-Max 8-Pin Low-Profile IC Socket (Mouser 575-113308)
- (3) Aries 8-Pin DIP Header (Mouser 535-08-600-11; Optional – see text)
- (4) Wima 470pF/100V, 5% Polypropylene Capacitor (C1, C2; Mouser 505-FKP2470/100/5)
- (2) Wima 100pF/100V, 5% Polypropylene Capacitor (C3; Mouser 505-FKP2100/100/5)
- (2) Black Gate PK-Series 220μF/4V Electrolytic Capacitor or Nichicon KZ-Series 220μF/25V Electrolytic Capacitor (C4; M. Percy)
- (4) 100Ω, 1% Vishay/Dale Metal Film Resistor (R1, R2; Mouser 71-RN60D-F-100)
- (2) 4.75k, 1% Vishay/Dale Metal Film Resistor (R3; Mouser 71-RN60D-F-4.75k)
- (2) 750k, 1% Vishay/Dale Metal Film Resistor (R4; Mouser 71-RN55D-F-750k)
- (4) Nichicon 100μF /25V FG-series Electrolytic Capacitor (C6-7, C10-11); M. Percy)
- (5) Wima – or – Roederstein 1.0μF /63V Polyester Capacitor (C5, C8-9, C12-13; Mouser #505-MKS21.0/63/5 or #75-MKT1817510064)
- (2) Neutrik NC3FD-L-B-1 Female, Chassis-mount XLR Connector (Mouser 568-NC3FD-L-1-B)

THAT 1646 Line Driver Board

(Quantities are for a 2-channel stereo board)

- (1) DH Labs D4 Prototype Board (Old Colony #PCBD-4)
- (2) THAT 1646 Balanced Line Driver IC (IC1; Mouser 887-1646P08-U)
- (2) Mill-Max 8-Pin Low-Profile IC Socket (Mouser 575-113308)
- (2) Aries 8-Pin DIP Header (Mouser 535-08-600-11; Optional – see text)
- (4) Nichicon ES-series 10μF/50V Non-Polar Electrolytic Capacitor (C1, C2; M. Percy)
- (4) Wima 100pF/100V, 5% Polypropylene Capacitor (C3, C4; Mouser 505-FKP2100/100/5)
- (4) Murata Ferrite Bead Inductor (L1, L2; Mouser 81-BL01RN1A1D2B)
- (4) Nichicon 100μF/25V FG-series Electrolytic Capacitor (C5-6, C9-10); M. Percy)
- (4) Wima or Roederstein 1.0μF /63V Polyester Capacitor (C7-8, C11-12; Mouser #505-MKS21.0/63/5 or #75-MKT1817510064)
- (2) Neutrik NC3MD-L-B-1 Male, Chassis-mount XLR Connector (Mouser 568-NC3MD-L-1-B)

ably high for most applications. Use very high quality non-polar electrolytic capacitors, such as the Nichicon ES series. A film capacitor would be ideal, but the physical size of a 10μF film cap, at any readily-available voltage, makes this an impractical choice. Use the Murata ferrite bead inductors and Wima FKP-2-series polypropylene capacitors for RFI protection, L1, L2, C3, and C4.

The 1646 incorporates a proprietary internal protection scheme which should be sufficient for the levels of ESD encountered in most normal field situations. It is possible, however, for any balanced output to be accidentally plugged into a microphone preamp with +48V phantom power. The 1646 datasheet shows an external scheme for protecting the chip from such hazards. If you think you need it, by all means use it.

I mentioned power supplies only in passing at the beginning of this article. Suffice to say, these chips will give their best performance with high-quality, regulated power supplies.

The THAT 1200 and 1646 balanced line receivers and drivers represent the state-of-the-art in active, IC-based balanced inputs and outputs. With a proper operating environment they offer excellent performance at very reasonable cost. *ax*

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VENDORS

Digi-Key Corporation
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530-470-8650
www.percyaudio.com
mpercy@pacbell.net

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Mansfield, TX 76063
800-346-6873 or 817-804-3888
www.mouser.com

Old Colony Sound Lab
PO Box 876

Peterborough, NH 03458-0876
888-924-9465 or 603-924-9464
www.audioXpress.com (click on "Shopping")
custserv@audioXpress.com

THAT Corporation
45 Sumner Street
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508-478-9200 (Voice)
508-478-0990 (FAX)
www.thatcorp.com

Parts Connexion
5109 Harvester Rd
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866-681-9602 or 905-681-9602 (Voice)
905-631-5777 (FAX)
www.partsconnexion.com

Champagne Fidelity: An Adventure *Par Excellence*

Jan Didden, while visiting historic wine country in northeast France with a group of fellow audio DIYers, was moved by this encounter with a French viticulteur.

By Jan Didden

Ofentimes a muting circuit for the output of a preamp is left out of DIY applications as an unnecessary or annoying part of the project. We met Cedric Bouchard after roaming around his residence in the village of Celles-Sur-Ource on the southern tip of the French Champagne region. This 30ish-looking Frenchman changed his slippers for something sturdier and invited us to follow him to his cave.

We had just spent a weekend at Stella Plage near Boulogne in Normandy enjoying a gathering of like minds at the European Triode Festival. One of our group had set up appointments for us to sample the champagne in the Reims area. Small vineyards usually don't open their gates to private customers, but instead cater to high-end restaurants and caves. Just a few minutes into the conversation with Cedric, my buddy whispered to me: "This guy would make a hell of an audiophile!," and I fully agreed with him. This is why.

After graduation, "this guy" left his sleepy village for the high-life of Paris, not being interested in the slightest in his father's small vineyard. But in Paris, between partying, he got into discussions about biological farming and ecological food products, and realized he could actually make a difference! Being the kind of person that puts his money where his mouth is, he returned to his father's place and started his own little champagne production. But he left all the rules and accepted ways behind him and started his own adventure, his own search for the

soul of the grape, letting Mother Nature do what she does best.

His production is tiny—a few thousand bottles from this plot, a few thousand from that, but very high quality. For example, he doesn't use the hydraulic presses everyone else does, but instead, keeps to the old ways: treading the grapes by foot! Less yield, of course, but much more character in the liquid—no additives of any kind, and no mixing of several years and several grape varieties to try to maintain a constant taste over time. His champagne is different from year to year.

He explains: "There are no bad years or good years, there are only different years. Every time one of my bottles is opened, a new adventure starts, with sometimes unexpected outcomes." He routinely opens a bottle from a lot for the very first time in the presence of other enthusiasts, because he wants to experience what people who buy his champagne experience when they open the bottle, and he wants to feel his own apprehension that is so necessary to maintain drive and ambition.

His father still operates his part of the vineyard in the traditional way. "How's your relation to your father?" I asked him at one point. He looked at me and said, smiling, "complicated." But, he adds, out of conflict come constructive ideas, more adventures.

Then he told me that a few years ago he got a disease that paralyzed him from the waist down. Incredibly, he overcame that and today he runs around without a trace of any limitations. But, he says, "it pro-

foundly changed my perspective on life. I am determined to live the life I want, to enjoy it to the fullest, to be my own man."

So, what does all this have to do with being an audiophile? What all of us recognized in this man—at a level we can only aspire to—were dedication and commitment to his craft. Isn't that also what the audio avocation is all about?

Determination to build the best amplifier, best speaker, best DAC you can. Even if it is not the world's best, it is your best effort, and something you can be proud of. Because it was you who made the decisions, it was you who solved all the myriad smaller and larger problems, you who did the painstaking parts selection and sourcing, and you who put it all together. Switching it on in the company of your friends, because you want to share that adventure and that apprehension and ultimately the enjoyment of having done something well.

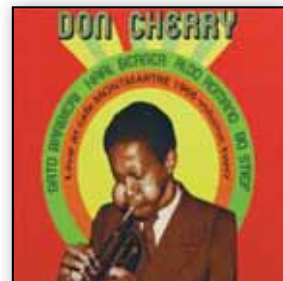
Maybe we should all remember this pride in accomplishment when we critique designs that do not conform to our own rules and ways, or when we voice disapproval on how a fellow audio DIYer solved a problem. Because it is not about the absolute measure of performance, it is about enjoying what you do, about being the best you can be. This hobby of ours is a way to enjoy life and to be proud of what we do, in a world replete with jobs and tasks that can be boring at best and dehumanizing at worst. Let us cultivate this avocation, let's promote it, and let's respect one another for it. *ax*

Jazz Track

By John Shand

Don Cherry | Live At Café Montmartre 1966 Vol. 2 | ESP/Planet ESP 4043

This magical album has two photographs in which Don Cherry's face radiates pure joy. They are as close as you'll get to photographs of his music. As well as helping formulate free jazz, Cherry was the preeminent pioneer of that music's most fruitful way of reinventing itself over the next 40 years. Even this 1966 band was a "world music" ensemble of sorts: the American trumpeter joined by Argentinean saxophonist Gato Barbieri, German vibraphonist Karl Berger, Danish bassist Bo Stief, and Italian drummer Aldo Romano. Cherry's other great innovation was suite-like constructions, interweaving composed and improvised sections. This album was recorded between the making of his two Blue Note masterpieces (*Complete Communion* and *Symphony for Improvisers*), and features material from both, including an exultant reading of the former's title suite.



The Alcohoticks | You, You | Jazzgroove JGR044

There's an audaciousness about the Alcohoticks' line-up of two guitars, drums, and no bass; a swagger as the trio marauds through country, rock, surf, and funk like three jazz Vikings, pillaging what they like; a brazenness in Evan Mannell's slapping the drums with a scorn some minors reserve for their kid brothers. You have to squint to see the jazz elements in this debut album: they're in the improvising skills and in the play-anything instrumental deftness. The surfaces loom largely as the sharp, hard ones of rock, and a collective sense of humor ensures a gift for pastiche. Aaron Flower and Ben Hauptmann both have stinging guitar sounds, whether hoeing down a country lick or snarling across odd-meter riffs, and Mannell's walloping abandon belies his nuclear precision.



The Antipodean Collective | Funcall | Extreme XCD 066

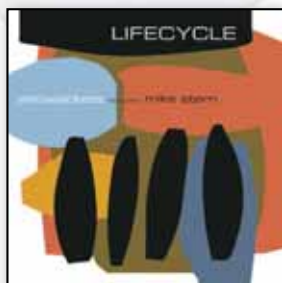
Imagine our language without articles, prepositions, or other niceties. It may briefly seem gibberish, then merely disjointed, before the meaning explodes amid provocative verbal juxtapositions. The Antipodean Collective's improvising language is a little like that. The thuds and needles of sound may initially seem arbitrary, but the listener's imagination can create its own links between notes. The starkness and drama come from the holes that trumpeter Scott Tinker, violinist John Rodgers, pianist Marc Hannaford, and especially drummer Ken Edie burn in the music's fabric. The results are as emotionally ambiguous as they are rhythmically and harmonically cryptic.

Plaintive cries and brutal thumps cohabit the same sentence, and the silence spilling around them gives each sound a sudden snap of surprise. The intensely interactive improvising has some otherworldly fusing of sounds.



Yellowjackets (featuring Mike Stern) | Lifecycle | Heads Up HUCD 3139

Mike Stern dons an honorary yellow jacket like it was made to measure. His slippery playing and cool sound slide easily into the smooth jazz-rock institution that is the Yellowjackets. "Cool" is the operative word, the approach bordering on clinical, especially with Marcus Baylor's drumming being so crisp that you half-expect the collective texture to snap at any moment. Standing out is the more impressionistic *Measure of a Man* by pianist and keyboard player Russell Ferrante, which features gorgeous soprano saxophone from Bob Mintzer and a lyrical solo from the composer. While Stern, Mintzer, and Ferrante add bursts of energy and snarls of intensity elsewhere, it is primarily the curling lines of electric bassist Jimmy Haslip that ensure a tide of humanity washes through the stainless steel and glass soundscape.



Kristin Berardi Band | The Kristin Berardi Band | Jazzhead HEAD093

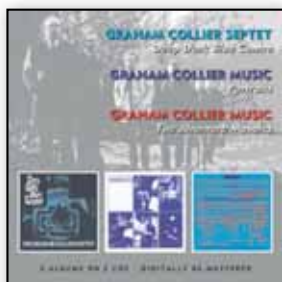
Singer Kristin Berardi has a beautiful voice, excellent pitch, and can improvise with conviction and imagination. Above all, she brings humility to her art, with a chink of vulnerability in the glistening armor of her vocal assurance. . . a chink that lets the listener feel what she feels, rather than merely observing it. She also has a fabulous band, with pianist Mike Nock, guitarist James Muller, bassist Mike Majkowski, and drummer Tim Firth.

But I am still not convinced by her songs. Ideally lyrics and music should complement the poetry implicit in each other. I find several of Berardi's songs to be too baldly confessional, and while some of the melodies are attractive, only occasionally do they become launchpads for hurling the extraordinary gift that is her voice into full musical and emotional flight.



Graham Collier | Deep Dark Blue Centre/Portraits/The Alternate Mosaics | BGO BGOCD822

Some reissues stand out amid the avalanche, sounding new-minted rather decades old, as with this trio of 1967–72 Graham Collier albums, now packaged on two CDs. The earliest, "Deep Dark Blue Centre," is, for the first time, on CD in stereo, highlighting the British composer's careful arranging of his pieces for trumpet/flugelhorn (Kenny Wheeler or Harry Beckett), Mike Gibbs' trombone, Dave Aaron's alto or flute, Karl Jenkins' baritone or oboe, Phil Lee's guitar, Collier's bass, and John Marshall's drums. Deft lines from flute or oboe emerge like sun-flecked highlights, and the brilliant soloists—especially Wheeler!—are supported by supple ensembles and fluid rhythms rather than locked-in blocks of sound—a lesson for most arrangers. The suite-like title track includes a captivating lament for trombone, guitar, flute, and bass.



FaitalPro's 15FX560 woofer is equipped with superior 700W nominal power, 99dB efficiency, and frequency range of 40-4000Hz. It contains a powerful magnetic motor, 3" voice coil, and linear excursion next to $\pm 10\text{mm}$. Also from FaitalPro, the 18XL1500 is a fully waterproof super woofer that can perform in mid-size 130/150 liter front loaded reflex or horn-loaded enclosures. It uses elastic electrical parameters and features tuning at 30 and 90Hz. The company's new 15XL1400 is a 15" subwoofer that reaches 1400W nominal power, and has a Q_t of 0.33. The 3FE20 full-range series, also from FaitalPro, feature 3", 80mm loudspeakers with pressed steel baskets and 91dB efficiency, resulting in reduced eddy currents. There are also two new compression drivers, the HF144 and HF204, which include a 65mm voice coil, Ketone polymer dome diaphragm, and 80W nominal power and 110dB dynamic range and 0.7-18kHz frequency response, respectively. To learn more, visit www.faitalpro.com.



Vincent Audio's C-60 is a high-end CD player with switchable vacuum tube and solid-state output stages, and fully balanced XLR outputs. Its chassis is made from solid aluminum plates, and it uses fully balanced circuitry with RCA unbalanced outputs and a coaxial digital output and optical digital output. Using a 6Z4 vacuum tube rectifier for the output stage power supply voltage, the C-60 has a frequency range of 10Hz-20kHz $\pm 0.5\text{dB}$. For more, go to www.wsdistributing.com.

The BM5A MkII, from **Dynaudio Acoustics** (www.dynaudioacoustics.com), is a near-field precision monitor offering longer excursions and louder, more accurate bass. It includes an added waveguide for less acoustical reflections, and three integrated analog room filters. Other features include handcrafted aluminum drivers, a soft-dome tweeter, two-channel 90W transient power, and an extended excursion 7" woofer.



Kicker's EB101M Premium in-ear stereo headset with microphone functions as both stereo speakers and a multimedia/microphone device (www.kicker.com). The headset is designed with high-quality audio in mind, and the music can switch over to a phone call with a single touch. Also, three sizes of noise-isolation silicon tips are included as well as a set of Comply™ memory foam tips.

OWI introduces a surface mount self-amplified speaker system with a matching passive 4 Ω speaker. The self-amplified speaker drives the other speaker, while combining the amplifier with the speaker maximizes damping control due to low source impedance. To learn more, visit www.owi-inc.com.





Onkyo's TX-SR608 THX-certified A/V receiver has a power amplifier section and a power boost from 90 to 100W. Also including a front HDMI input, the TX-SR608 features Audyssey DSX dimensional sound processing, and the ability to upscale all video sources to 1080p. Onkyo presents other home theater-in-a-box systems that decode lossless master audio bitstreams, and DACs that are resistant to clock jitter. For more information, please visit www.onkyousa.com.

The BG500 bass combos, from **TC Electronic** (www.tcelectronic.com), feature a fully integrated bass tuner that automatically mutes when activated, allowing for swift, silenced tuning. Both the BG500 115 and the BG500 210 include three user memories, an on-board tuner, and 500W or 1000W of peak power.



Harman International (www.harman.com) has released several new sets of headphones. The 316s are lightweight, with comfortable ear clips. The 370s are high-end in-ear headphones with high sensitivity, wide dynamic range, and high SPL levels, as well as a flight adapter, inline microphone, and a carrying case for a companion phone. The 390 NC high-end noise cancelling in-ear headphones are engineered to play loud and long, with accurate, uncolored sound. Finally, the 390 NC, which is Bluetooth optional, contains two lengths of single-sided cable.

The iconic Reveal studio monitor, from **Tannoy** (www.Tannoy.com), has returned, and features two new monitors: the 501a and 601a. The 501a is an active monitor with a 60W integrated amplifier and an LED indicator and rear-mounted volume control. The 601a is a compact speaker with extended low-frequency response, a 1" soft dome tweeter, and a 90W total power amp unit. The Reveal range all feature front-firing reflex bass, wideband tweeters, optimized response, and balanced XLR and unbalanced jack connectors.



CONTRIBUTORS

Gary Galo ("New York AES 2009" and "THAT's Balanced Line Drivers and Receivers") is Audio Engineer at The Crane School of Music, SUNY Potsdam, where he also teaches courses in music literature. A contributor to AAC since 1982, he has authored over 230 articles and reviews on audio technology, music, and recordings. He has been the Sound Recording Reviews Editor of the *ARSC Journal* (Association for Recorded Sound Collections) since 1995, was co-chair of the ARSC Technical Committee from 1996 to 2004, and has given numerous presentations at ARSC conferences (www.arsc-audio.org). Mr. Galo is also a frequent book reviewer for *Notes: Quarterly Journal of the Music Library Association*, has written for the *Newsletter of the Wilhelm Furtwangler Society of America*, and is the author of the "Loudspeakers" entry in *The Encyclopedia of Recorded Sound in the United States*, 1st edition.

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

John Shane ("Jazz Track") is a CD reviewer for *Australian HiFi* and *Home Theatre Technology*.



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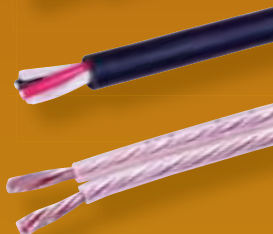


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