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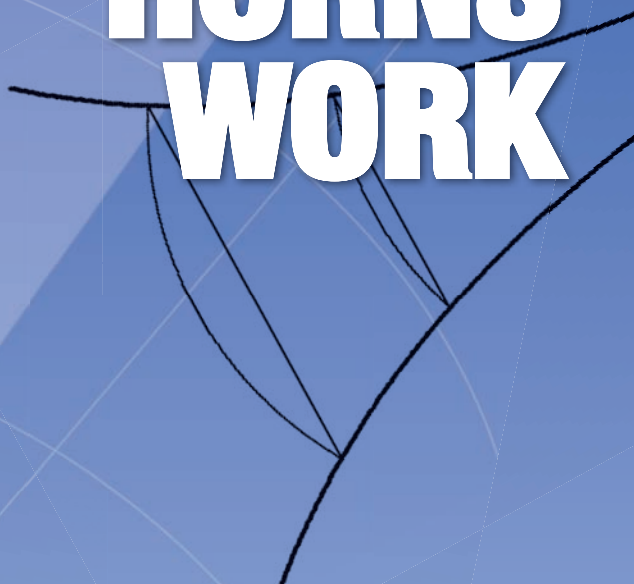

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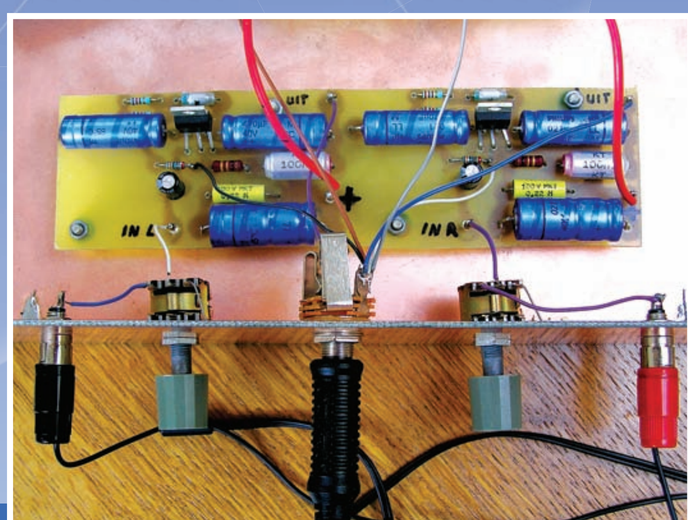

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Horn Theory: An Introduction, Part 1

This author presents a two-part introduction to horns—their definition, features, types, and functions.

This article deals with the theory of acoustical horns, as it applies to loudspeakers. It reviews the basic assumptions behind classical horn theory as it stands, presents the different types of horns, and discusses their properties. Directivity control, wave-front shapes, and distortion are also discussed.

In this article, I try to keep the math simple, and, where it is required, I explain or illustrate the meaning of the equations. The focus is on understanding what is going on in a horn. The practical aspects of horn design are not treated here.

TERMINOLOGY

The article includes the following terminology:

Impedance: Quantity impeding or reducing flow of energy. Can be electrical, mechanical, or acoustical.

Acoustical Impedance: The ratio of sound pressure to volume velocity of air. In a horn, the acoustical impedance will increase when the cross-section of the horn decreases, as a decrease in cross section will limit the flow of air at a certain pressure.

Volume Velocity: Flow of air through a surface in m^3/s, equals particle velocity times area.

Throat: The small end of the horn, where the driver is attached.

Mouth: The far end of the horn, which radiates into the air.

Driver: Loudspeaker unit used for driving the horn.

c: The speed of sound, 344m/s at 20° C.

ρ_0: Density of air, 1.205 kg/m^3.

f: Frequency, Hz.

ω: Angular frequency, radians/s, $\omega = 2\pi f$.

k: Wave number or spatial frequency, radians/m, $k = \frac{\omega}{c} = \frac{2\pi f}{c}$.

S: Area.

p: Pressure.

Z_A: Acoustical impedance.

j: Imaginary operator, $j = \sqrt{-1}$.

THE PURPOSE OF A HORN

It can be useful to look at the purpose of the horn before looking at the theory. Where are horns used, and for what?

Throughout the history of electroacoustics, there have been two important aspects:

- Loading of the driver
- Directivity control

You would also think that increasing the output would be one aspect of horns, but this is included in both. Increasing the loading of the driver over that of free air increases efficiency and hence the output, and concentrating the sound into a certain solid angle increases the output further.

Loading of the Driver. The loudspeaker, which is a generator of pressure, has an internal source impedance and drives an external load impedance. The air is the ultimate load, and the impedance of air is low, because of its low density.

The source impedance of any loudspeaker, on the other hand, is high, so there will be a considerable mismatch between the source and the load. The result is that most of the energy put into a direct radiating loudspeaker will not reach the air, but will be converted to heat in the voice coil and mechanical resistances in the unit. The problem is worse at low frequencies, where the size of the source will be small compared to a wavelength and the source will merely push the medium away. At higher frequencies, the radiation from the source will be in the form of plane waves that do not spread out. The load, as seen from the driver, is at its highest, and the system is as efficient as it can be.

If you could make the driver radiate plane waves in its entire operating range, efficient operation would be secured at all frequencies. The driver would work into a constant load, and if this load could be made to match the impedances of the driver, resonances would be suppressed. This is because the driver is a mechanical filter, which needs to be terminated in its characteristic impedance, ideally a pure resistance. If the driver is allowed to radiate plane waves, resistive loading is secured.

The easiest way to make the driver radiate plane waves is to connect it to a long, uniform tube. But the end of the tube will still be small compared to a wavelength at low frequencies. To avoid reflections, the cross section of the tube must be large compared to a wavelength, but, at the same time, it must also be small to fit the driver and present the required load. To solve this dilemma, you need to taper the tube.

When you do this, you can take radiation from the driver in the form of plane waves and transform the high pressure, low velocity vibrations at the throat into low pressure, high velocity vibrations that can efficiently be radiated into the air. Depending on how the tube flares, it is possible to present a load to the driver that is constant over a large frequency range.

Directivity Control. The directivity of a cone or dome diaphragm is largely uncontrolled, dictated by the dimensions of the diaphragm, and heavily dependent on frequency, becoming sharper and sharper as frequency increases. You can solve this problem by using multiple driving units and digital signal processing, but a far simpler and cheaper way to achieve predictable directivity control is to use a horn. The walls of the horn will restrict the spreading of the sound waves, so that sound can be focused into the areas where it is needed, and kept out of areas where it is not.

Directivity control is most important in sound reinforcement systems, where a large audience should have the same distribution of low and high frequencies, and where reverberation and reflections can be a problem. In a studio or home environment, this is not as big a problem.

As the art and science of electroacoustics has developed, the focus has changed from loading to directivity control. Most modern horns offer directivity control at the expense of driver loading, often presenting the driver with a load full of resonances and reflections. **Figure 1** compares the throat impedance of a typical constant directivity horn (dashed line) with the throat impedance of a tractrix horn (solid line)[1]. The irregularities above 8kHz come from higher order modes.

FUNDAMENTAL THEORY

Horn theory, as it has been developed, is based on a series of assumptions and simplifications, but the resulting equations can still give useful information about the behavior. I will review the assumptions later, and discuss how well they hold up in practice.

The problem of sound propagation in horns is a complicated one, and has not yet been rigorously solved analytically. Initially, it is a three-dimensional problem, but solving the wave equation in 3D is very complicated in all but the most elementary cases. The wave equation for three dimensions (in Cartesian coordinates) looks like this[2]

$$\frac{\partial^2\phi}{\partial t^2} - c^2\left(\frac{\partial^2\phi}{\partial x^2} + \frac{\partial^2\phi}{\partial y^2} + \frac{\partial^2\phi}{\partial z^2}\right) = 0 \qquad (1)$$

This equation describes how sound waves of very small (infinitesimal) amplitudes behave in a three-dimensional medium. I will not discuss this equation, but only note that it is not easily solved in the case of horns.

In 1919, Webster[3] presented a solution to the problem by simplifying equation 1 from a three-dimensional to a one-dimensional problem. He did this by assuming that the sound energy was uniformly distributed over a plane wavefront perpendicular to the horn axis, and considering only motion in the axial direction. The result of these simplications

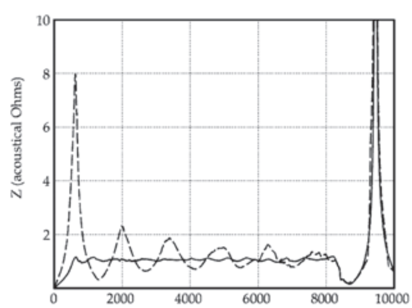

FIGURE 1: Throat impedance of a typical constant directivity horn (—) and a tractrix horn (—)1.

is the so-called "Webster's Horn Equation," which can be solved for a large number of cases:

$$\frac{d^2\phi}{dx} + \frac{d\ln S}{dx}\frac{d\phi}{dx} - k^2\phi = 0 \qquad (2)$$

where

$k = \frac{2\pi f}{c}$, the wave number or spatial frequency (radians per meter),
ϕ is the velocity potential (see appendix for explanation), and S is the cross-sectional area of the horn as a function of x.

The derivation of equation 2 is given in the appendix. You can use this equation to predict what is going on inside a horn, neglecting higher order effects, but it can't say anything about what is going on outside the horn, so it can't predict directivity. Here are the assumptions equation 2 is based on[4,5]:

1. Infinitesimal amplitude: The sound pressure amplitude is insignificant compared to the steady air pressure. This condition is easily satisfied for most speech and music, but in high power sound reinforcement, the sound pressure at the throat of a horn can easily reach 150-170dB SPL. This article takes a closer look at distortion in horns due to the nonlinearity of air later, but for now it is sufficient to note that the distortion at home reproduction levels is insignificant.
2. The medium is considered to be a uniform fluid. This is not the case with air, but is permissible at the levels (see 1) and frequencies involved.
3. Viscosity and friction are neglected. The equations involving these quantities are not easily solved in the case of horns.
4. No external forces, such as gravity, act on the medium.
5. The motion is assumed irrotational.
6. The walls of the horn are perfectly rigid and smooth.
7. The pressure is uniform over the wave-front. Webster originally considered tubes of infinitesimal cross-section, and in this case propagation is by plane waves. The horn equation does not require plane waves, as is often assumed. But it requires the wave-front to be a function of x alone. This, in turn, means that the center of curvature of the wave-fronts must not change. If this is the case, the horn is said to admit one-parameter (1P) waves[6], and according to Putland[7], the only horns that admit 1P waves are the uniform tube, the parabolic horn with cylindrical wave-fronts, and the conical horn. For other horns, you need the horn radius to be small compared to the wavelength.

Because the horn equation is not able to predict the interior and exterior sound field for horns other than true 1P horns, it has been much criticized. It has, however, been shown[8, 9] that the approximation is not as bad as you might think in the first instance. Holland[10] has shown that you can predict the performance of horns of arbitrary shape by considering the wave-front area expansion instead of the physical cross-section of the horn. I have also developed software based on the same principles, and have been able to predict the throat impedance of horns with good accuracy.

SOLUTIONS

This section presents the solution of equation 2 for the most interesting horns, and looks at the values for throat impedance for the different types. You can calculate this by solving the horn equation, but this will not be done in full mathematical rigor in this article.

The solution of equation 2 can, in a general way, be set up as a sum of two functions u and v:

$$\phi = Au + Bv \qquad (3)$$

where A and B represent the outgoing (diverging) and reflected (converging) wave, respectively, and u and v depend on the specific type of horn.

In the case of an infinite horn, there is no reflected wave, and B = 0. This article first considers infinite horns, and presents the solutions for the most common types[11]. The solutions are given in terms of absolute acoustical impedance,

$$\frac{\rho_o c}{S_t};$$

you can find the specific throat impedance (impedance per unit area) by multiplying by S_t, the throat area, and the normalized throat resistance by multiplying by $\frac{S_t}{\rho_o c}$.

THE PIPE AND THE PARABOLIC HORN

Both these horns are true 1P horns. The infinite pipe of uniform cross-section acts as a pure resistance equal to

$$z_A = \frac{\rho_o c}{S_t} \qquad (4)$$

An infinite, uniform pipe does not sound very useful. But a suitably damped, long pipe (plane wave tube) closely approximates the resistive load impedance of an infinite pipe across a wide band of frequencies, and is very valuable for testing compression drivers[12, 13]. It presents a constant frequency independent load, and as such acts like the perfect horn.

The parabolic horn is a true 1P horn if it is rectangular with two parallel sides, the two other sides expanding linearly, and the wave-fronts are concentric cylinders. It has an area expansion given as $S = S_t x$. The expression for throat impedance is very complicated, and will not be given here.

The throat impedances for both the uniform pipe and the parabolic horn are given in **Fig. 2**. Note that the pipe has the best, and the parabolic horn the worst, loading performance of all horns shown.

CONICAL HORN

The conical horn is a true 1P horn in

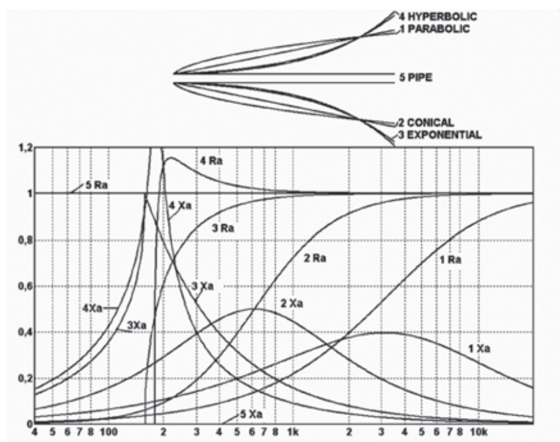

FIGURE 2: Throat acoustical resistance r_A and reactance x_A as a function of frequency for different horn types.

spherical coordinates. If you use spherical coordinates, the cross-sectional area of the spherical wave-front in an axisymmetric conical horn is $S = \Omega(x+x_0)^2$, where x_0 is the distance from the vertex to the throat, and Ω is the solid angle of the cone. If you know the half angle θ (wall tangent angle) of the cone,

$$\Omega = 2\pi(1 - \cos\theta). \quad (5)$$

In the case where you are interested in calculating the plane cross-sectional area at a distance x from the throat,

$$S(x) = S_t\left(\frac{x + x_0}{x0}\right)^2. \quad (6)$$

The throat impedance of an infinite conical horn is

$$z_A = \frac{\rho_o c}{S_t}\left(\frac{k^2x_0^2 + jkx_0}{1 + k^2x_0^2}\right). \quad (7)$$

You should note that equation 7 is identical to the expression for the radiation impedance of a pulsating sphere of radius x_0.

The throat resistance of the conical horn rises slowly (**Fig. 2**). At what frequency it reaches its asymptotic value depends on the solid angle Ω, being lower for smaller solid angle. This means that for good loading at low frequencies, the horn must open up slowly.

As you will see later, a certain minimum mouth area is required to minimize reflections at the open end. This area is larger for horns intended for low frequency use (it depends on the wavelength), which means that a conical horn would need to be very long to provide satisfying performance at low frequencies. As such, the conical expansion is not very useful in bass horns. Indeed, the conical horn is not very useful at all in applications requiring good loading performance, but it has certain virtues in directivity control.

EXPONENTIAL HORN

Imagine you have two pipes of unequal cross-sectional areas S_0 and S_2, joined by a third segment of cross-sectional area S_1, as in **Fig. 3**. At each of the discontinuities, there will be reflections, and the total reflection of a wave passing from S_0 to S_2 will depend on S_1. It can be shown that the condition for least reflection occurs when

$$S_1 = \sqrt{S_0S_2}. \quad (8)$$

This means that $S_1 = S_0k$ and $S_2 = S_1k$, thus $S_2 = S_0k^2$. Further expansion along this line gives for the nth segment, $S_n = S_0k^n$, given that each segment has the same length. If $k = e^m$, and n is replaced by x, you have the exponential horn, where the cross-sectional area of the wave-fronts is given as $S = S_te^{mx}$. If you assume plane wave-fronts, this is also the cross-sectional area of the horn at a distance x from the throat.

The exponential horn is not a true 1P-horn, so you cannot exactly predict its performance. But much information can be gained from the equations.

The throat impedance of an infinite exponential horn is

$$z_A = \frac{\rho_o c}{S_t}\left(\sqrt{1 - \frac{m^2}{4k^2}} + j\frac{m}{2k}\right) \quad (9)$$

When m = 2k or $f = \frac{mc}{4\pi}$, the throat resistance becomes zero, and the horn is said to cut off. Below this frequency, the throat impedance is entirely reactive and is

$$z_A = j\frac{\rho_o c}{S_t}\left(\frac{m}{2k} - \sqrt{\frac{m^2}{4k^2} - 1}\right). \quad (10)$$

The throat impedance of an exponential horn is shown in **Fig. 2**. Above the cutoff frequency, the throat resistance rises quickly, and the horn starts to load the driver at a much lower frequency than the corresponding conical horn. In the case shown, the exponential horn throat resistance reaches 80% of its final value at 270Hz, while the conical horn reaches the same value at about 1200Hz.

An infinite horn will not transmit anything below cutoff, but it's a different matter with a finite horn, as you will see later.

You should note that for an exponential horn to be a real exponential horn, the wave-front areas, not the cross-sectional areas, should increase exponentially. Because the wave-fronts are curved, as will be shown later, the physical horn contour must be corrected to account for this.

HYPERBOLIC HORNS

The hyperbolic horns, also called hyperbolic-exponential or hypex horns, were first presented by Salmon[14], and is a general family of horns given by the wave-front area expansion

$$S = S_t\left(\cosh\frac{x}{x_0} + T\sinh\frac{x}{x_0}\right)^2. \quad (11)$$

T is a parameter that sets the shape of the horn (**Fig. 4**). For T = 1, the horn is an exponential horn, and for $T \rightarrow \infty$ the horn becomes a conical horn.

x_0 is the reference distance given as $x_o = \frac{c}{2\pi f_c}$ where f_c is the cutoff frequency. A representative selection of hypex contours is shown in **Fig. 4**.

Above cutoff, the throat impedance of an infinite hyperbolic horn is

$$z_A = \frac{\rho_o c}{S_t}\left(\frac{\sqrt{1 - \frac{1}{\mu^2}}}{1 - \frac{1 - T^2}{\mu^2}} + j\frac{\frac{T}{\mu}}{1 - \frac{1 - T^2}{\mu^2}}\right), \quad (12)$$

and below cutoff, the throat impedance is entirely reactive and is

$$z_A = j\frac{\rho_o c}{S_t}\left(\frac{\frac{1}{\mu}\sqrt{\frac{1}{\mu^2} - 1}}{1 - \frac{1 - T^2}{\mu^2}}\right) \quad (13)$$

where

μ is the normalized frequency, $\mu = \frac{f}{f_c}$.

The throat impedance of a hypex horn with T = 0.5 is shown in **Fig. 2**. The throat impedance of a family of horns with T ranging from 0 to 5 is shown in **Fig. 5**.

Exponential and hyperbolic horns have much slower flare close to the

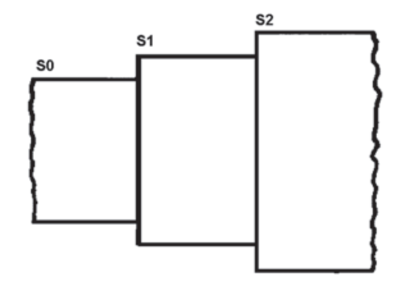

FIGURE 3: Joined pipe segments.

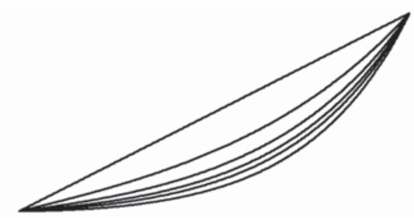

FIGURE 4: A family of representative hypex contours, T = 0 (lower curve), 0.5, 1, 2, 5, and infinite (upper curve).

throat than the conical horn, and thus have much better low frequency loading. When T < 1, the throat resistance of the hyperbolic horn rises faster to its asymptotic value than the exponential, and for $T < \sqrt{2}$ it rises above this value right above cutoff. The range 0.5 < T < 1 is most useful when the purpose is to improve loading. When T = 0, there is no reactance component above cutoff for an infinite horn, but the large peak in the throat resistance may cause peaks in the SPL response of a horn speaker.

Due to the slower flaring close to the throat, horns with low values of T will also have somewhat higher distortion than horns with higher T values.

WHAT IS CUTOFF?

Both exponential and hyperbolic horns have a property called cutoff. Below this frequency, the horn transmits nothing, and its throat impedance is purely reactive. But what happens at this frequency? What separates the exponential and hyperbolic horns from the conical horn that does not have a cutoff frequency?

To understand this, you first must look at the difference between plane and spherical waves[10]. A plane wave propagating in a uniform tube will not have any expansion of the wavefront. The normalized acoustical impedance is uniform and equal to unity through the entire tube.

A propagating spherical wave, on the other hand, has an acoustical impedance that changes with frequency and distance from the source. At low frequencies and small radii, the acoustical impedance is dominated by reactance. When kr = 1—i.e., when the distance from the source is $\frac{\lambda}{2\pi}$—the reactive and resistive parts of the impedance are equal, and above this frequency, resistance dominates.

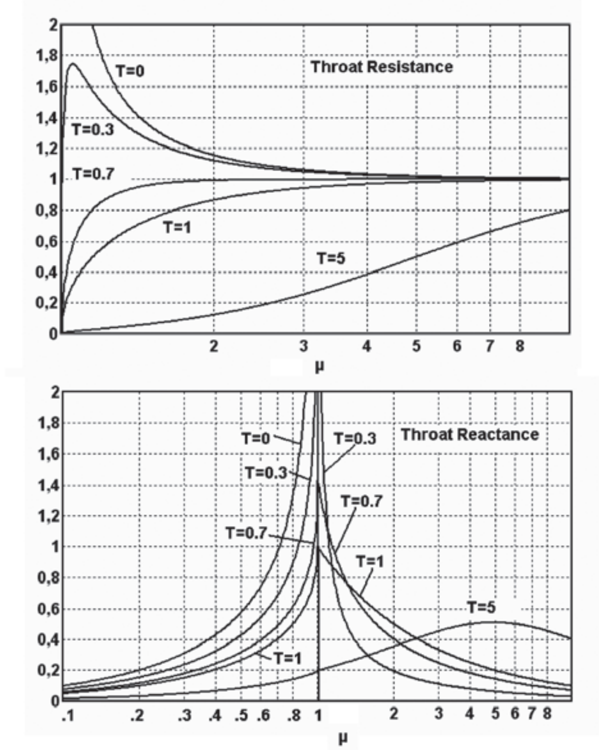

FIGURE 5: Throat impedance of a family of hypex horns.

The difference between the two cases is that the air particles in the spherical wave move apart as the wave propagates; the wave-front becomes stretched. This introduces reactance into the system, because you have two components in the propagating wave: the pressure that propagates outward, and the pressure that stretches the wave-front. The propagating pressure is the same as in the non-expanding plane wave, and gives the resistive component of the impedance. The stretching pressure steals energy from the propagating wave and stores it, introducing a reactive component where no power is dissipated. You can say that below kr = 1, there is reactively dominated propagation, and above kr = 1 there is resistive dominated propagation.

If you apply this concept to the conical and exponential horns by looking at how the wave-fronts expand in these two horns, you will see why the cutoff phenomenon occurs in the exponential horn. You must consider the flare rate of the horn, which is defined as (rate of change of wave-front area with distance)/(wave-front area).

In a conical horn, the flare rate changes throughout the horn, and the point where propagation changes from reactive to resistive changes with frequency throughout the horn.

In an exponential horn, the flare rate is constant. Here the transition from reactive to resistive wave propagation happens at the same frequency throughout the entire horn. This is the cutoff frequency. There is no gradual transition, no frequency dependent change in propagation type, and that's why the change is so abrupt.

FINITE HORNS

For a finite horn, you must consider both parts of equation 3. By solving the horn equation this way[3, 15], you get the following results for pressure and volume velocity at the ends of a horn:

$p_m = ap_t + bU_t$ (14)
$U_m = fp_t + gU_t$ (15)

where p and U denote the pressure and volume velocity, respectively, and the subscripts denote the throat and mouth of the horn. You can now find the impedance at the throat of a horn, given that you know a, b, f, and g:

$$Z_t = \frac{gZ_m - b}{a - fZ_m} \qquad (16)$$

where Z_m is the terminating impedance at the mouth.

The expressions for a, b, f, and g are quite complicated, and are given by Stewart[15] for the uniform tube, the conical, and the exponential horn.

You see that the value of mouth impedance will dictate the value of the throat impedance. As explained previously, there will usually be reflections at the mouth, and depending on the phase and magnitude of the reflected wave, it may increase or decrease the throat impedance. A horn with strong reflections will have large variations in throat impedance.

Reflections also imply standing waves and resonance. To avoid this, it is important to terminate the horn correctly, so that reflections are minimized. This will be discussed in the next section.

It's interesting to see what effect the length has on the performance of a horn. **Figure 6** shows the throat impedance of 75Hz exponential horns of different lengths, but the same mouth size. As the horn length increases, the throat resistance rises faster to a useful value, and the peaks in the throat impedance become more closely spaced.

Finite horns will transmit sound below their cutoff frequency. This can be explained as follows: the horn is an acoustical transformer, transforming the high impedance at the throat to a low impedance at the mouth. But this applies only above cutoff. Below cutoff there is no transformer action, and the horn only adds a mass reactance.

An infinite exponential horn can be viewed as a finite exponential horn terminated by an infinite one with the same cutoff. As you have seen, the throat resistance of an infinite exponential horn is zero below cutoff, and the throat resistance of the finite horn will thus be zero. But if the impedance present at the mouth has a non-zero resistance below cutoff, a resistance will be present at the throat. This is illustrated in **Fig. 7**, where a small exponential horn with a mouth three times larger than its throat is terminated by an infinite pipe (a pure acoustical resistance). The acoustical resistance present at the throat below cutoff approaches that of the pipe alone, one third of the value above cutoff.

The same is true for any mouth termination. As long as there is a resistive impedance present at the mouth below cutoff, power can be drawn from the horn.

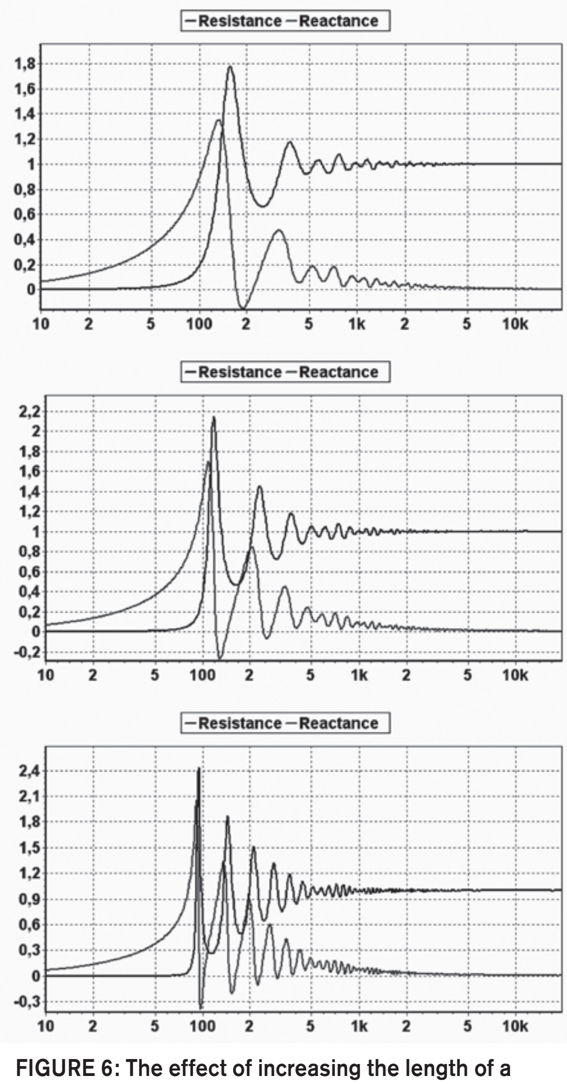

FIGURE 6: The effect of increasing the length of a 75Hz exponential horn with krm = 0.5. The lengths are (top to bottom) 50, 100, and 200cm.

TERMINATION OF THE HORN

I have briefly mentioned that there can be reflections from the mouth of a horn. The magnitude of this reflection depends on frequency and mouth size.

Consider a wave of long wavelength[16]. While it is progressing along a tube, it occupies a constant volume, but when it leaves the tube, it expands into an approximate hemispherical shape (**Fig. 8**). The volume thus increases, the pressure falls, increasing the velocity of air inside the tube, pulling it out. This produces an impulse that travels backwards from the end of the tube, a reflection.

It's a different matter at higher frequencies. The relative volume of a half-wavelength of sound is much smaller, and the resulting volume increase is less, producing fewer reflections (**Fig. 9**).

You may ask what the optimum size of the end of the tube is, to minimize reflection in a certain frequency range. This has been investigated since the early 1920s[2, 5, 16, 17, 18] and has led to the general assumption that if the mouth circumference of an exponential horn is at least one wavelength at the cutoff frequency of the horn, so that $kr_m \geq 1$, the reflections will be negligible. r_m is the radius of the mouth.

The effect of different mouth sizes is shown in **Fig. 10**, where the throat impedance of a 100Hz exponential horn is

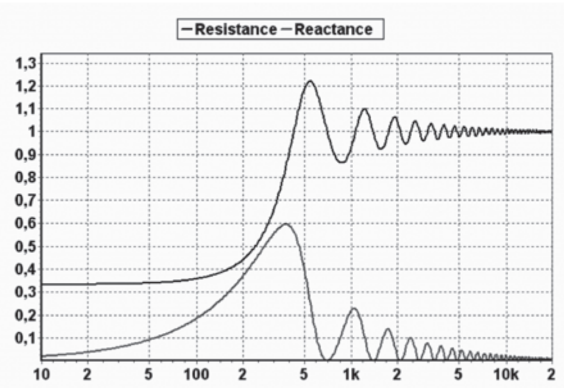

FIGURE 7: Finite exponential horn terminated by an infinite pipe.

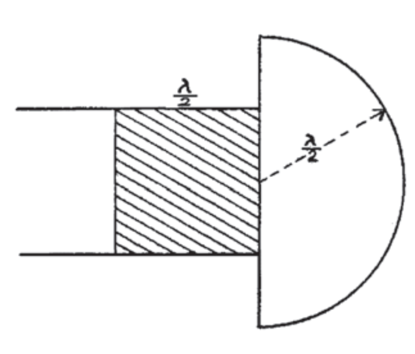

FIGURE 8: Waves with long wavelength at the end of a tube[16].

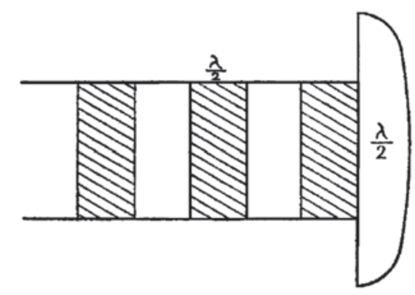

FIGURE 9: Waves with short wavelength at the end of a tube[16].

shown. The throat impedance is calculated assuming plane wave-fronts, and using the impedance of a piston in an infinite wall as termination. The mouth sizes correspond to kr_m = 0.23, 0.46, 0.70, and 0.93. You can see that for higher values of kr_m, the ripple in the throat impedance decreases.

When kr_m is increased beyond 1, however, the ripple increases again, as shown in **Fig. 11**. This led Keele to investigate what the optimum horn mouth size would be[19]. For a horn termination simulated by a piston in an infinite baffle, he found the optimum kr_m to be slightly less than 1, the exact value depending on how close to cutoff the horn is operating. His findings were based on the plane wave assumption, which, as you will see in the next section, does not hold in practice.

As a historical side note, Flanders also discovered increased mouth reflections for kr_m larger than 1 for a plane wave exponential horn[18] in 1924.

If the same horn is calculated assuming spherical waves, there is no obvious optimum mouth size. If you consider the throat impedance of two 100Hz horns with kr_m = 0.93 and 1.4, assuming spherical wave-fronts and the same mouth termination as before, you can see that the ripple decreases, not increases, for higher values of kr_m (**Fig. 12**).

The reason may be that the wave-front expansion of a horn where the cross-section is calculated as $S = S_t e^{mx}$ will have a flare rate that decreases toward the mouth. This is because the wave-fronts bulge (**Fig. 13**). The areas of the curved wave-fronts are larger than those of the plane ones, and the distance between them is also larger. But the distance between successive curved wave-fronts increases faster than their areas, so the outer parts of the horn will have a lower cutoff. The required kr_m for optimum termination becomes larger, and it increases as the horn is made longer.

These considerations are valid for exponential horns. What, then, about hyperbolic and conical horns? Hyperbolic horns with T < 1 will approximate the exponential horn expansion a certain distance from the throat, and the mouth termination conditions will be similar to those for an exponential horn. Conical horns show no sign of having an optimum mouth size. As length and mouth size increase, simulations show that the throat impedance ripple steadily decreases, and the horn approaches the characteristics of an infinite horn.

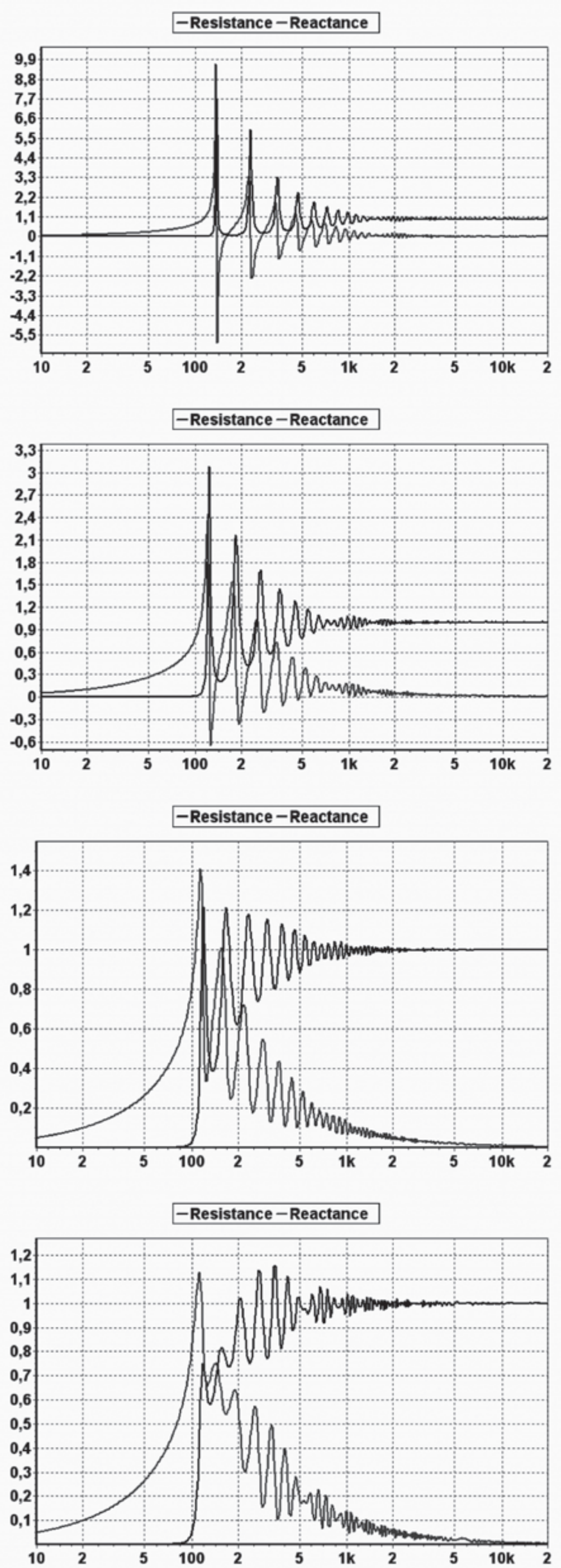

FIGURE 10: Throat impedance of finite horns[10], kr_m = (top to bottom) 0.23, 0.46, 0.70, and 0.93.

In conclusion, you may say that the mouth area of a horn can hardly be made too large, but it can easily be made too small. kr_m in the range 0.7-1 will usually give smooth response for bass horns, while midrange and tweeter horns will benefit from values ≥ 1.

Another factor you need to consider is the termination at the throat. If there is a mismatch between the driver and the horn, the reflected waves traveling from the mouth will again be reflected when they reach the throat, producing standing waves in the horn.

CURVED WAVE-FRONTS

By logical reasoning, the assumption that the wave-fronts in a horn are plane cannot be true. If it was so, the speed of sound along the horn walls would need to be greater than the speed of sound along the axis. This cannot be the case, and the result is that the wave-front on the axis must gain on that at the horn walls, so the wave-fronts will define convex surfaces. In a circular conical horn the wave-fronts are spherical with centers at the apex. In the uniform tube, the wave-fronts are plane. In both cases the wave-fronts cut the axis and the walls at right angles. It is, therefore, logical to assume that this will happen in other horn types, too.

WAVE-FRONTS IN HORNS

In 1928, Hall conducted a detailed investigation of the sound field inside horns[4, 20] showing how the wave-fronts curve in an exponential and a conical horn. The wave-fronts in a 120Hz exponential horn at the cutoff frequency are shown in **Fig. 14**, where you can see that the wave-fronts are very nearly normal to the walls.

It's a different matter at 800Hz (**Fig. 15**). At a certain distance from the throat, the pressure wave-fronts become seriously disturbed. Hall attributes this to reflections at the outer rim of the mouth that are more powerful than at the center, because the discontinuity is greater. Another explanation[8] is that higher order modes (see Part 2) will distort the shape of the amplitude wave-fronts. This is also most evident in **Fig. 15**. In a flaring horn, higher order modes will not appear at the same frequencies throughout the horn. Close to the throat, where the radius is small, they will appear at fairly high frequencies, but closer to the mouth they will appear at lower frequencies.

Conical horns do not look any better than exponential horns. Hall also investigated a large conical horn, 183cm long, throat diameter 2cm, and mouth diameter 76cm. Simulations show that the horn does not have significant mouth reflections, because the throat impedance is close to that of an infinite horn. Still the amplitude wave-fronts are seriously disturbed, even close to the throat, which does not happen in an exponential horn (**Fig. 16**). You can see two nodal lines, each about halfway between the horn wall and the axis. This is a result of higher order modes, and can be predicted.

The frequency where Ra = Xa—i.e., where the throat acoustical resistance and reactance are equal—is about 1kHz for this horn. This indicates that higher order modes are a problem in conical horns even below the frequency where it has useful loading properties.

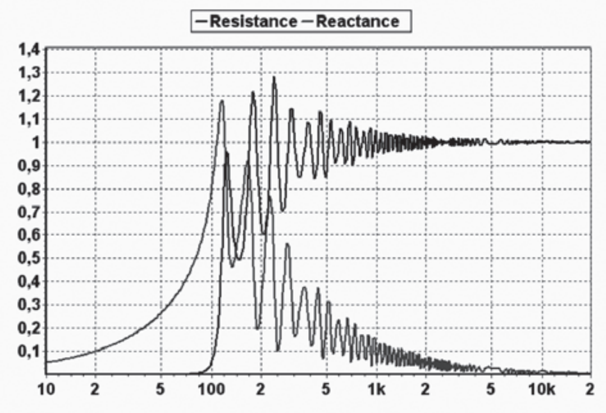

FIGURE 11: Increasing ripple for oversized mouth, kr_m = 1.4.

TRACTRIX HORN

The tractrix is a kind of horn generated by the revolution of the tractrix curve around the x-axis. The equation for the tractrix curve is given as

$$x = r_m \ln \frac{r_m + \sqrt{r_m^2 - r_x^2}}{r_x} - \sqrt{r_m^2 - r_x^2} \qquad (17)$$

where

r_m is the mouth radius, usually taken as $\frac{\lambda_c}{2\pi} = \frac{c}{2\pi f_c}$, where f_c is the horn cutoff frequency, and r_x is the radius of the horn at a distance x from the horn mouth (**Fig. 17**).

Because the radius (or cross-section) is not a function of x, as in most other horns, the tractrix contour is not as straightforward to calculate, but it should not pose any problems.

The tractrix horn expands faster than the exponential horn close to the mouth, as you will see in Part 2.

The tractrix curve was first employed for horn use by P.G.A.H. Voigt, and patented in 1926[21]. In more recent times it was popularized by Dinsdale[22], and most of all by Dr. Bruce Edgar[23, 24]. The main assumption in the tractrix

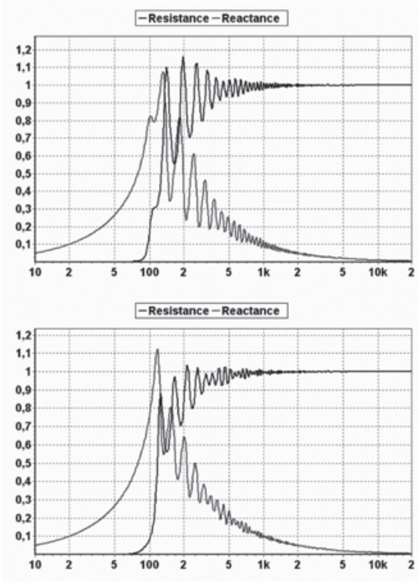

FIGURE 12: Throat impedance of horns with krm = 0.93 and 1.4, assuming spherical wave-fronts.

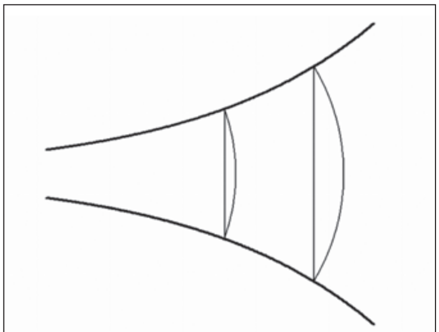

FIGURE 13: Comparing plane and curved wave-fronts in an exponential horn.

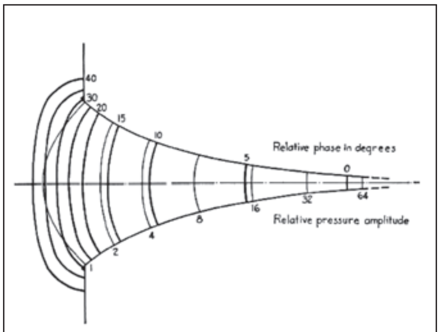

FIGURE 14: Wave-fronts in an exponential horn at 120Hz[4].

APPENDIX
DERIVATION OF THE WEBSTER HORN EQUATION

This derivation is based on the infinitesimal amplitude, one-parameter plane wave assumption, as discussed in Part 1.

Consider a flaring horn as shown in **Fig. A**, where dx is the short axial length between two plane wave-fronts of area S. The volume of this element is Sdx, where S is given as an arbitrary function of x. Fluid (air) will flow into this element from one side, and out of it on the other side, due to the passage of sound waves. The change in the mass of the fluid in this volume is

$$-\frac{\partial(\rho S)}{\partial t}dx$$

with dx not changing with time.

The particle velocity of the fluid moving along the x-axis through the element is u, and the difference in the mass of fluid entering one plane and leaving the other, is

$$\rho\frac{\partial(uS)}{\partial x}dx$$

this is ρ (the density of the medium) times the change in volume velocity, uS, with x. Because the fluid is continuous, these two quantities must be equal, so

$$\rho\frac{\partial(uS)}{\partial x} = -\frac{\partial(\rho S)}{\partial t}$$

Expand both sides to get

$$\rho\left(u\frac{\partial S}{\partial x} + S\frac{\partial u}{\partial x}\right) = -\left(\rho\frac{\partial S}{\partial t} + S\frac{\partial \rho}{\partial t}\right)$$

Now introduce the concept of velocity potential, which you can consider as similar to electric potential along a resistive conductor. If this conductor has a resistance R per unit length, the resistance of a small length ∂x is R∂x. If a current I flows through the conductor, the voltage drop (electrical potential) across ∂x is
$\partial U = -RI\partial x$.
Setting R = 1, you have

$$I = -\frac{\partial U}{\partial x}.$$

Acoustically, you may say that the velocity potential replaces U, and the particle velocity replaces I. Thus $u = -\frac{\partial\phi}{\partial x}$.

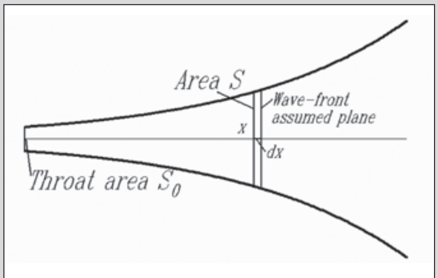

FIGURE A: Plane wave propagation in a horn.

You also have the relation that

$$\frac{\partial\rho}{\partial t} = \rho_0\frac{\partial s}{\partial t} = \frac{\rho_0}{c^2}\frac{\partial^2\phi}{\partial t^2},$$

where s is the condensation of the medium, and ρ_0 is the static density of the medium. For infinitesimal amplitudes, $\rho = \rho_0$, and $\frac{\partial A}{\partial t} = 0$, because the area at a given value of x is independent of time. After substitution, you have

$$\frac{\partial^2\phi}{\partial x^2} + \left[\frac{1}{A}\frac{\partial A}{\partial x}\right]\frac{\partial\phi}{\partial x} - \frac{1}{c^2}\frac{\partial^2\phi}{\partial t^2} = 0.$$

Because $\frac{1}{S}\frac{\partial S}{\partial x} = \frac{\partial 1nS}{\partial x}$, you can write this as

$$\frac{\partial^2\phi}{\partial x^2} + \frac{\partial 1nS}{\partial x}\frac{\partial\phi}{\partial x} - \frac{1}{c^2}\frac{\partial^2\phi}{\partial t^2} = 0$$

This is the fundamental horn equation for infinitesimal amplitudes. If you have simple harmonic motion (a sine or cosine wave of a single frequency), you can write $\phi = \phi_1 \cos \omega t$, which gives $\partial^2\phi/\partial t^2 = -\omega^2\phi$, $\omega = 2\pi f$. By using this substitution, and remembering that $k = \omega/c$, you get

$$\frac{d^2\phi}{dx} + \frac{d1nS}{dx}\frac{d\phi}{dx} - k^2\phi = 0,$$

which is the most convenient form of the horn equation. ■

horn is that the sound waves propagate through the horn as spherical wave-fronts with constant radius, r_m, which also is tangent to the walls at all times (**Fig. 18**).

For this requirement to hold, the wave-front must be spherical at all frequencies, and the velocity of the sound must be constant throughout the horn.

A theory of the tractrix horn was worked out by Lambert[25]. The throat impedance of a horn was calculated using both a hemisphere and a piston as radiation load, and the results compared to measurements. It appeared that the wave-front at the mouth was neither spherical nor plane. Also, directivity measurements showed increased beaming at higher frequencies. This means that the tractrix horn does not present a hemispherical wave-front at the mouth at all frequencies. It does come close at low frequencies, but so does almost every horn type.

The throat impedance of a 100Hz tractrix horn, assuming wave-fronts in the form of flattened spherical caps and using the radiation impedance of a sphere with radius equal to the mouth radius as mouth termination, is shown in **Fig. 19**.

Next month Part 2 will continue this in-depth look at various horn types. aX

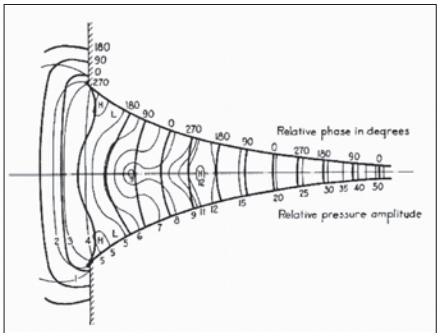

FIGURE 15: Wave-fronts in an exponential horn at 800Hz[4].

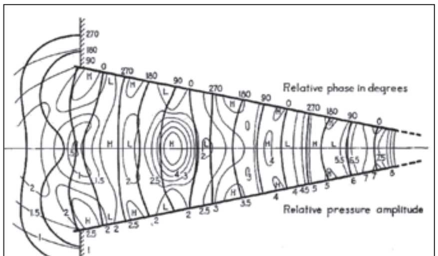

FIGURE 16: Wave-fronts in a conical horn at 800Hz[4].

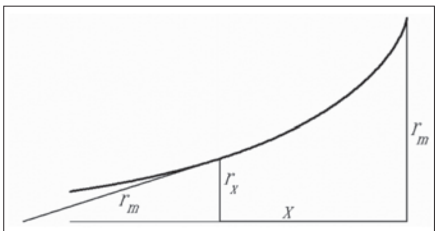

FIGURE 17: Dimensions of the tractrix horn.

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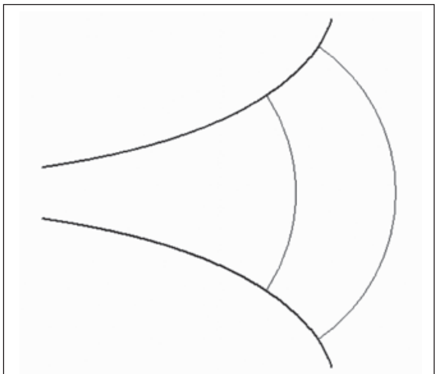


FIGURE 18: Assumed wave-fronts in a tractrix horn.

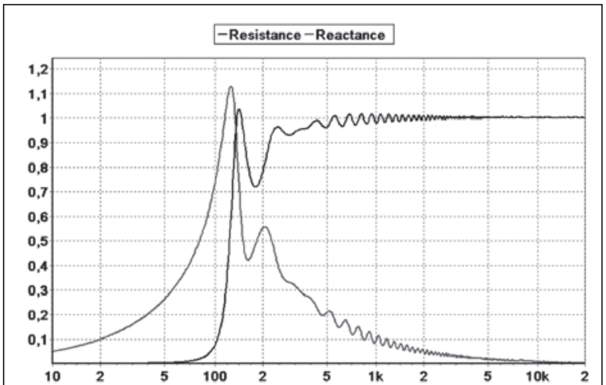


FIGURE 19: Throat impedance of a tractrix horn.

Amp6

Nothing unconventional here. . . just a good, solid amp design, using that venerable workhorse, the 6V6.

It is surprising how you sometimes think that something is gone, but it is not. For example, as long and as active as I have been in electronics, I was unaware that tube amplifiers are still very popular and, even more, that there are many tubes of different kinds on the market today. I asked a new student not long ago, "What kind of work do you do?" He replied, "I service tube type amplifiers." I told him that he was working on some *old* stuff. He said, "Mr. Laughlin, most are brand new, there are older ones also."

Man! As dumb as it sounds, I was amazed when he informed me that they are a really going concern today, and that there is a famous magazine devoted to them—*audioXpress*. I could hardly believe it! I had been pretty neglectful and non-observing beyond belief of part of a world I had cut my teeth on.

I immediately brought myself up on the present. How many loving memories I have of the bygone days when I thought tubes would forever be the only active devices. I remember the first transistor I had ever seen and thought, "Man, what is this dingy little thing? How can anybody think these will ever be useful or good for anything?" Wrong again! (it gets to be almost record setting).

Well, I got over my shock and decided to build a tube amp. It would be my first one in 49 years! I did, and this is it (**Photo 1**). I absolutely love it! I hope all of you will also.

I decided to use circuit boards for this project. My student friend told me one thing he almost hated about most tube amps with circuit boards was that you must remove the circuit board to replace almost any part. I had a *good* idea! I decided to configure the boards so that all parts are mounted on the bottom (**Photo 2**).

PHOTO 1: The 6V6 tube-based amplifier.

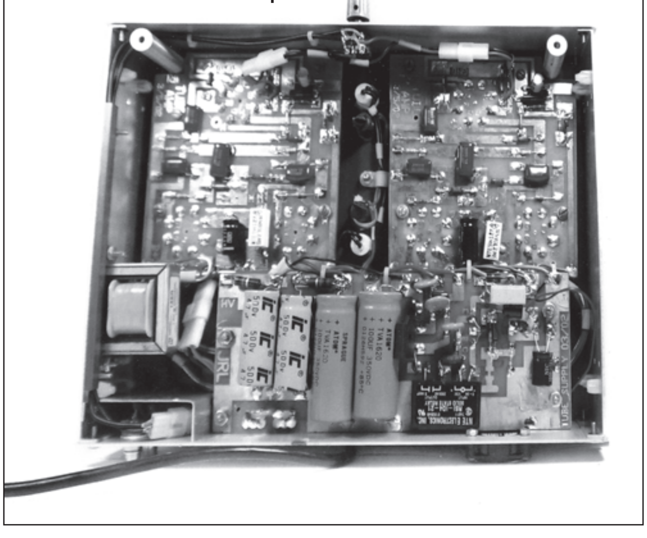
Some of the parts need a bit of machining to construct, which any machine can do easily and cheaply.

CHARACTERISTICS

I had these considerations in mind as I devised this project:

1. Stronger than normal 6V6 circuit boards.
2. A layout that permits total access to all components without needing to remove the boards from the chassis.
3. A method of providing some degree of cooling for the tubes.
4. A wiring scheme that makes for easy removal of the boards from the chassis arrangement if ever necessary.
5. A physical cabinet design a bit more pleasant to behold than "normal."
6. Increased tube life.

PHOTO 2: Inside the amp.



THE CABINET

First, obtain the circuit boards. The cabinet I used is just about perfect for the job. I decided to hide the transformers under a decorative cover. The first thing you should do is to construct this cover and mount it on the cabinet (**Figs. 1** and **2**). Take care that the rear edge of the cover aligns vertically with the rear edge of the chassis. The three mounting bolts along the front edge and the two rear mounting posts will secure it substantially to the cabinet.

Next, prepare mounting holes for the transformers under the cover, with outputs left and right and the power centered. Space the outputs approximately 1″ from the centered power. They should be oriented with the cores aligned in a front-to-back direction. Cut holes in the cabinet at appropriate locations to accept the transformer connecting wires and an additional hole on each side of the power transformer to accept the speaker terminal wires. Do not mount them at this time.

MOUNTING THE CIRCUIT BOARDS

To determine the exact location for the

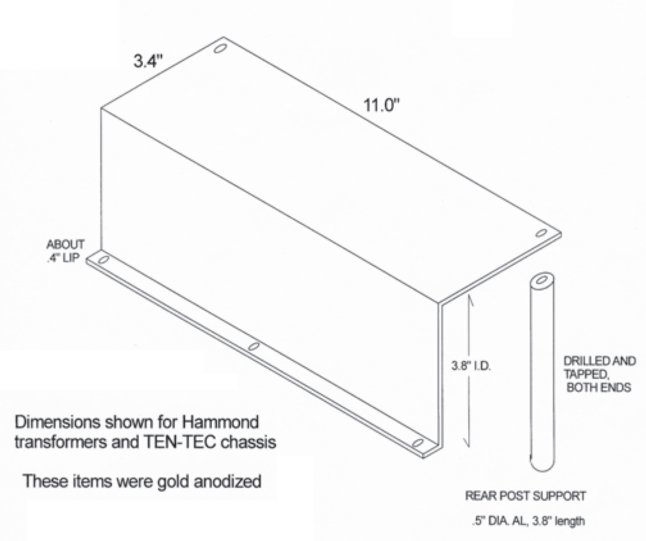


FIGURE 1: Transformer cover.

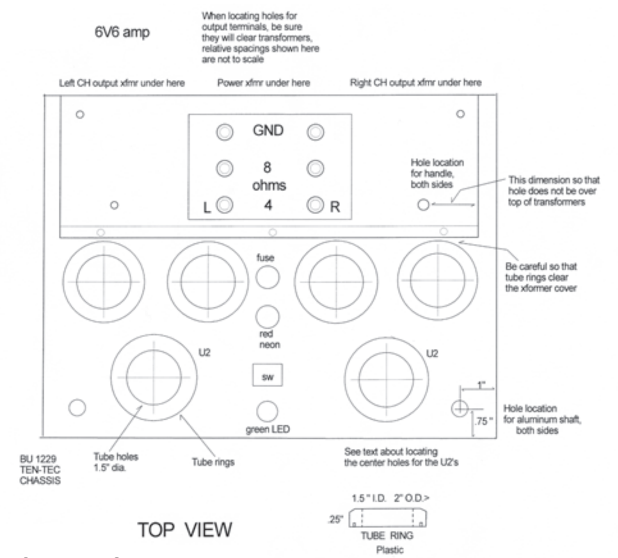


FIGURE 2: Chassis layout.

PHOTO 3: Top view.

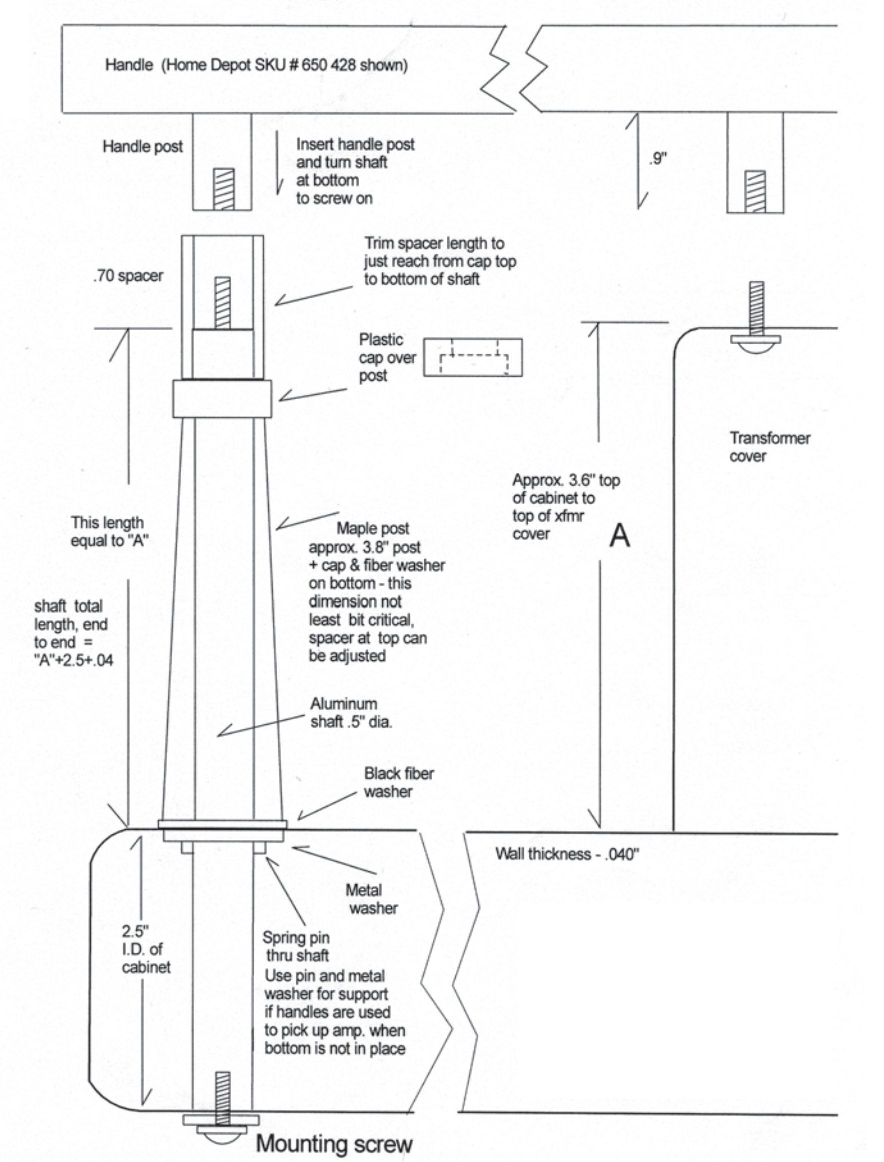


FIGURE 3: Handle front shaft and post assembly—two each.

6V6 boards, place this board inside the cabinet, making sure the top is against the cabinet top. Position the board so that the 6SL7 end of it just touches the front panel and the side of the board is approximately 0.35″ from the side of the cabinet. Carefully drill a small hole through the cabinet top using the center hole of the 6SL7 tube socket as the guide.

Place a pin through both holes to lock the board in place. Next, drill a small hole through the cabinet using one of the 6V6 tube socket centers. Pin these two holes together. These pins will hold the board in place while you drill the five mounting holes for the board into the cabinet top and the center hole for the other 6V6 tube socket.

After this is done for both 6V6 boards, remove them and place the power supply board inside the cabinet with its back edge touching the rear panel. Be certain the top of the board is against the cabinet top panel. Approximately center the board with respect to the two sides, being certain the mounting holes for this board are not in an interfering position with any transformer holes.

Using one of the board mounting holes in the board as a guide, drill the hole in the cabinet. Pin the board and cover together with a drill bit and drill the next mounting hole. Pin the board and cover together with two drill bits and drill the other two mounting holes in the cover. I used flat-head bolts for mounting the posts to the cabinet because of the transformer bottoms.

When mounting this power board, use posts ½″ longer than the 6V6 posts to elevate it above 6V6 boards. There is hardly any space to spare, so be careful about this layout, or components will not fit!

The front pilot lamp is L1, the green DC lamp. Behind it is the push-on-push off switch. Farther back is the fuse and red neon "operational" lamp. Drill and cut all the necessary holes, 1.5″ for the tubes. You can mount the tube rings using very small self-tapping screws.

Carefully drill the forward handle-mounting holes on top of the chassis. Use a small drill long enough to drill the top and also to reach down to drill the bottom screw holes. This should be done on a drill press with the chassis

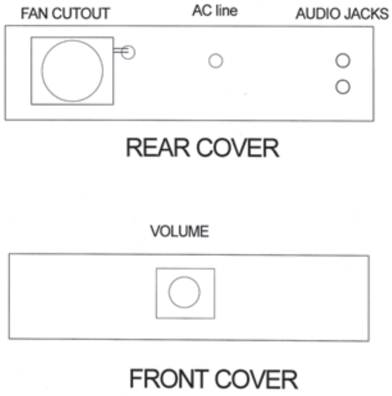


FIGURE 4: Chassis cutouts—front and rear.

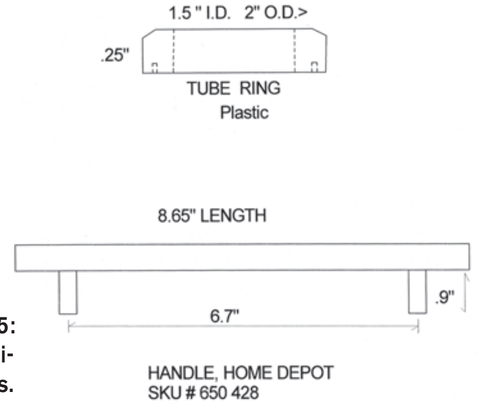


FIGURE 5: Handle dimensions.

held very accurately against the table to ensure accurate vertical alignment of the hole pairs. Then enlarge the top hole to fit the handle support rod.

With the handle support rods mounted and the handles bolted to them, you can judge where to drill the handle holes in the transformer cover. It is vital that the front handle support rods be trimmed to a length that places their tops exactly on the same horizontal plane as the top of the transformer cover. I suggest you mount the handles close enough to each side of the transformer cover because it will be difficult to manipulate the mounting screws underneath the cover if you do not. Notch one corner of each tube circuit board—one board on the left and the other board on the right, or both if desired—to accommodate the front handle supports.

I drilled holes in the transformer cover for the output jacks so that they straddle each side of the power transformer. I made a cover plate with the appropriate nomenclature engraved upon it as shown in **Photo 3**. I squeezed in the filter inductor alongside the power supply board.

Figures 3–6 give some details on the handle and post items and rear and front panels. I went to a considerable amount of trouble to use the small molex type connectors for the three boards, as shown in **Figs. 7** and **8**. Of course, the cabling can be brought out left or right. I really enjoy ease-of-removal of boards and components.

To ensure proper grounding of the chassis parts, I carefully connected ground leads to each and to the power supply ground. I mounted the double volume control on the front panel and connected input jacks and board inputs with small shielded cables

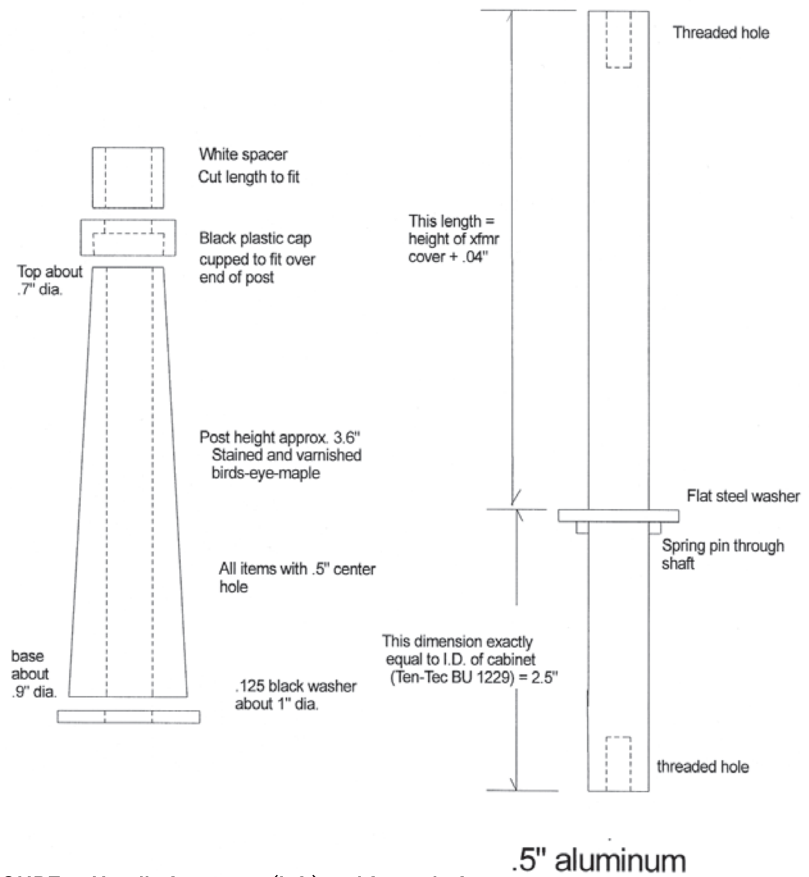


FIGURE 6: Handle front post (left) and front shaft.

using the molex connectors to the rear panel.

The power cord and fan are mounted on the rear panel and wired with molex connectors to their respective places (**Photo 4**). The hot wire from the power cord goes straight to the fuse as shown in the schematics. Air from the fan blows up and around the tubes through the clearance of their mounting holes in the cabinet top, helping to cool them.

LOADING THE BOARDS

6V6 boards: On the 6V6 board there are ten wires exiting the circuitry. The schematic describes this. I like to run these wires tightly alongside the edge of the board in one bundle and secure them with a wire-wrap fastened through a hole drilled appropriately near one corner of the board. This makes for a neater wiring arrangement and gives really good stress relief for the soldered joints of the wires. Soldering the parts onto the boards is, in my opinion, straightforward and simple.

Power Supply Board: Suggested wiring connections involving the six molex connectors for this board was my prefer-

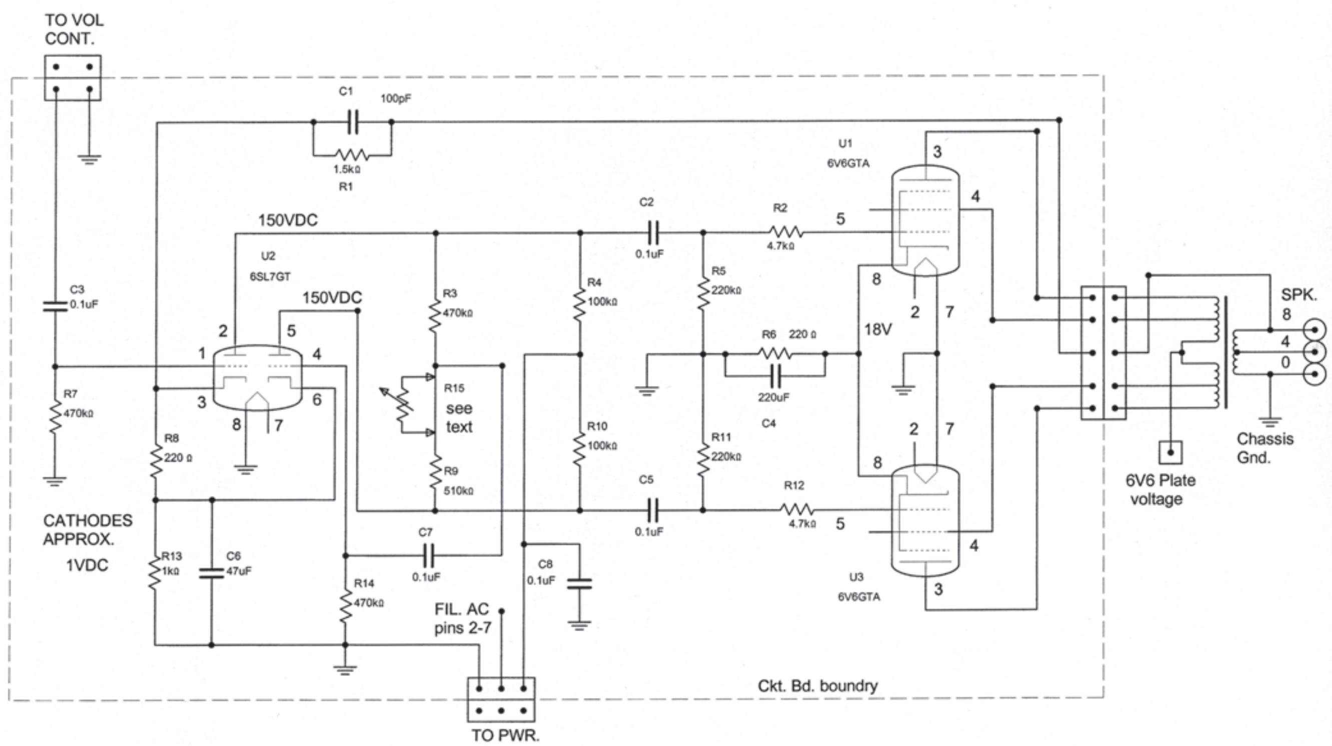


FIGURE 7: Amp circuit.

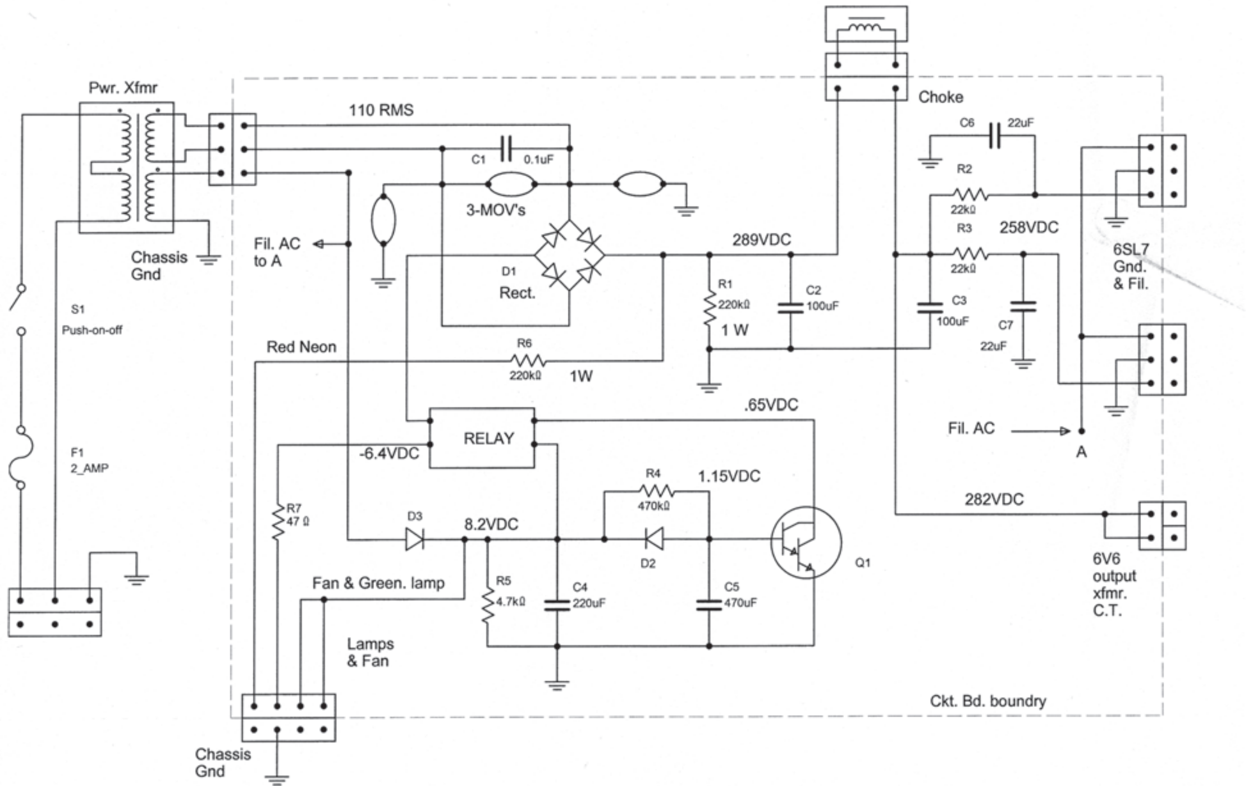


FIGURE 8: Power supply circuit.

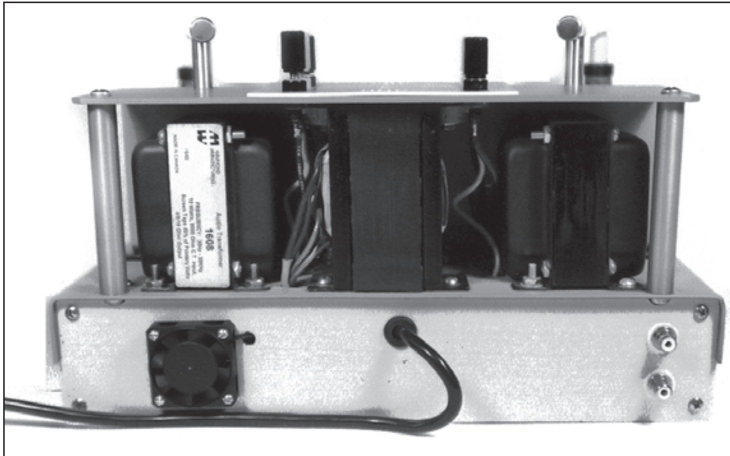


PHOTO 4: Rear view.

ence. You may have different and better ideas. As above, all parts are "surface" mounted. If desired, any lead that connects to ground can be fed through to the top cover also. Refer to the schematic for wiring details.

THE AMPLIFIER

I dug through some of my older amplifier books, mainly *The Recording and Reproduction of Sound* by Oliver Read, published in 1952, that I have had all these years and devised the circuit based mostly upon the McIntosh type of amp. A friend of mine told me, "Laughlin, this amp looks just about like some of the others I have seen, did you design it?" "Well," I told him, "as long as 6V6s and 6SL7s have been around, you try to design me an amp using them that you will not find a near duplicate of somewhere!" So this amp is not different or new in design but based on proven good results. One difference I have not seen to date (I believe) is the U2b connection. It is wired with negative feedback through R9 from output (pin 5) to input (pin 4).

This connection is similar to an op amp connected for an inverting gain of 1. I adjusted the resistor ratio between R3 and R9 to give U2b the gain of 1 to exactly match the output of pin 2. Variable R15 placement is provided for if you desire it.

POWER SUPPLY

The power supply is very standard with the addition of the Q1 delay circuit with a relay for applying the plate voltage to the tubes. D3 rectifies the filament voltage and C4 smoothes it.

When the power switch is turned on, R4 and C5 delay the base voltage rise to Q1; with the values shown, the delay is approximately 20 seconds. During this time the filaments are warming up. When Q1 turns on, the relay connects the high voltage diode bridge to ground through R7 and plate voltage is applied. I used R7 to ease the surge current through the relay contacts when charging C2.

Upon turning the power switch off, D2 causes C5 to discharge faster through R5. I could have designed this circuit much fancier, but it works as perfectly as ever needed, as-is.

I used three MOVs to ensure against impulse (more commonly referred to as "surge") damage to the diode rectifier bridge. Note that the red neon lamp indicates when the high voltage is applied, and the green LED lamp shows that the power switch is turned on. aX

TABLE 1
PARTS LIST 6V6 TUBE BOARDS

RESISTORS		
ALL RESISTORS ¼W UNLESS NOTED		
R1	1.5kΩ	
R2,12	4.7k	
R3,7,14	470k	
R4,10	100k	
R5,11	220k	
R6	220 1W	
R8	220	
R9	510K	
R13	1k	
CAPACITORS		
C1	100pF	50WV film
C2,3,5,7,8	0.1F	400WV film
C4	220µF	50WV electrolytic
C6	47µF	25WV electrolytic
TUBES		
U2	6SL7	
U1,U3	6V6	
SOCKETS		
Octal sockets	3 each	ceramic, PC mount
CONNECTORS		
1	5 circuit receptacle	Molex 3061052
1	5 circuit plug	Molex 3062051
1	3 circuit receptacle	Molex 3061032
1	3 circuit plug	Molex 3062032
1	2 circuit receptacle	Molex 3061023
1	2 circuit plug	Molex 3062023
STANDOFFS		
5 each	½″ 4-32 THREAD	
2 each	#4 fiber washers (use with standoffs next to 6V6s)	
HANDLE		
Home Depot	SKU – 650 428	

TABLE 2
PARTS LIST POWER SUPPLY BOARD

Parts available from Radio Shack unless otherwise stated		
RESISTORS		
ALL RESISTORS ¼W UNLESS NOTED		
R1	220kΩ, 1W	
R2,3	22k	
R4	470k	
R5	4.7k	
R6	1k	
CAPACITORS		
C1	0.1µF	400WV Film
C2,3	100µF	450WV Electrolytic
C4	220µF	16WV electrolytic
C5	470µF	16WV electrolytic
MOV		
M1,M2,M3	Metal-oxide-varistor V275LA2 JAMECO	275WV
DIODES		
D1	Full-wave bridge rectifier 50WV	
D2	1N914	
D3	1N4003 "	
TRANSISTOR		
Q1	MPSA13 or equivalent Darlington	
SOLID STATE RELAY		
SS1	NTE RSI-D-21 From various NTE distributors	
MECHANICAL RELAY – Radio Shack 275-249 or OEG OMI-SS-212D		
STANDOFFS		
4 each	1½″ 6-32 thread	

TABLE 3
PARTS LIST 6V6 TUBE BOARDS

Parts available from Radio Shack unless otherwise noted.	
TRANSFORMERS	
2 each	Hammond 1608 tube output
1 each	Hammond 261M6 plate-filament
1 each	Hammond 156R 1.5H filter inductor
PILOT LAMPS	
1 each	Red neon
1 each	Green 12V incandescent
SWITCH	
1 each	Push-on/push-off
FUSE	
1 each	2A slow-blow
1 each	fuse holder, panel mount
FAN	
1 each	12V DC 19/16 × 1/2 size
AUDIO INPUT JACKS	
1 pair	RCA phono jacks, insulated and color coded
AC LINE CORD	
1 each	Standard three-wire
VOLUME CONTROL	
1 each	Dual 10kΩ audio taper
SPEAKER JACKS	
2 each	Yellow banana binding post
2 each	Red binding post
2 each	Black binding post
CABINET	
TEN - TEC BU 1229 1-800-231-8842 for distributors 12-2.5-9″ blue - enclosuresales@tentec.com	

CIRCUIT BOARD BASICS

There are no holes to drill for components, which are surface mounted on the bottom of the boards. There are five feedthrough holes on all boards marked "FT." Drill these holes and use a piece of wire for the feedthrough. Refer to the layout guides in **Figs. A-F**.

Each tube socket on the board has a center-point indicated with a pad. Drill a small hole through each and drill holes for the tube socket pins. Also drill holes for the mounting posts, marked "M"—five on the 6V6 boards and four on the power supply board. For the 6V6 boards use a #4 bolt that has a small diameter head because the clearance for screw heads on the 6V6 boards is not great. If you must use a screw head with a diameter too large, place an insulating washer under it.

For the power board, bolt head diameter is not important. The top of this board is dedicated only to ground conductor and with the four feedthroughs it enhances the overall conduction. If desired, you can use a single sideboard (bottom only) for the power board and eliminate this extra ground layer and the feedthroughs. Note that there is one non-ground conductor on the top of the 6V6 board, the filament trace.

You may install two types of relays on the power board:

1. solid-state as shown on the layout
2. mechanical, which is mounted in the eight pad holes below the solid-state outline.

If you use the mechanical relay, a two sideboard is best, but with some small thru-hole wire jumpers a single sideboard is OK. With a thru-hole type configuration, you will need to remove the board to replace it, if ever. The two inner-most pins shown going to ground are not necessary and can be snipped off if desired.

Note: I cut an extra 6V6 shaped board, without the component pattern but with the ground plane and filament trace side and epoxied it onto the top of the component board, resulting in a double thickness board that is much stronger. I admit this was not needed, but at the time I liked the idea.

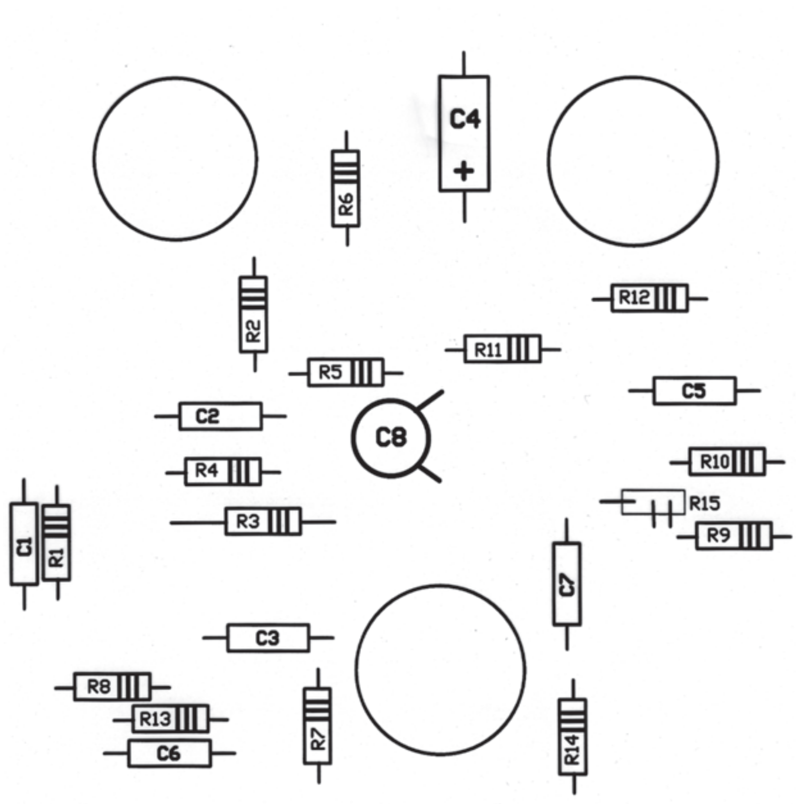


FIGURE A: Bottom overlay.

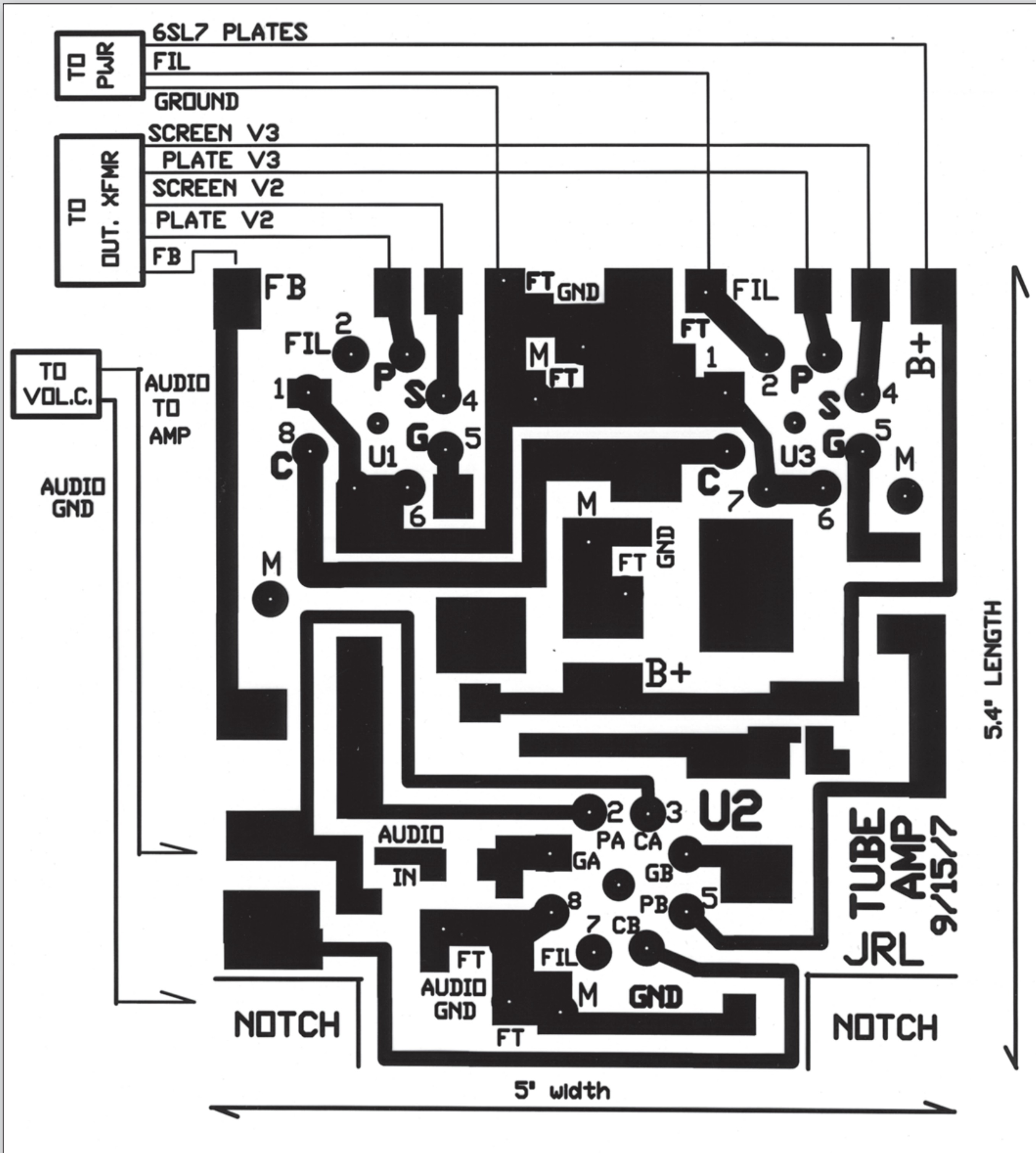


FIGURE B: Bottom layer, with parts surface-mounted on bottom, except tube sockets, which are on top.

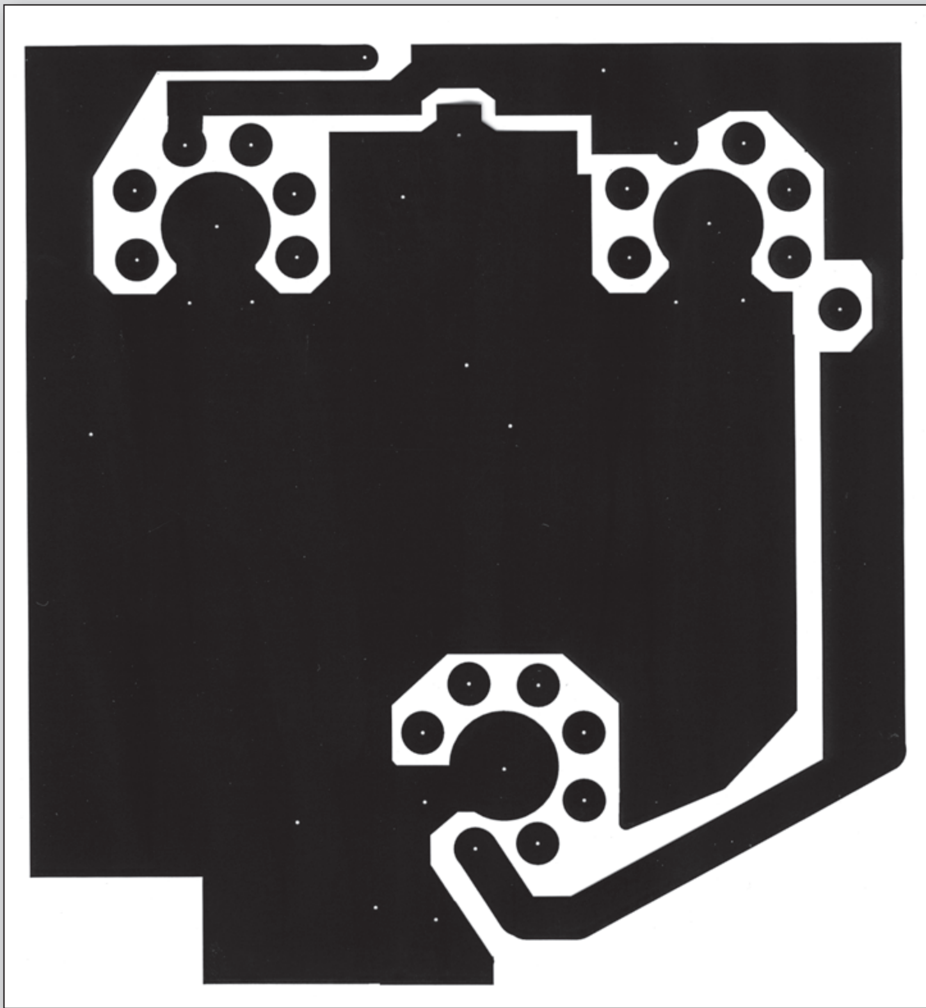


FIGURE C: Top layer.

FIGURE D: Power supply top layer.

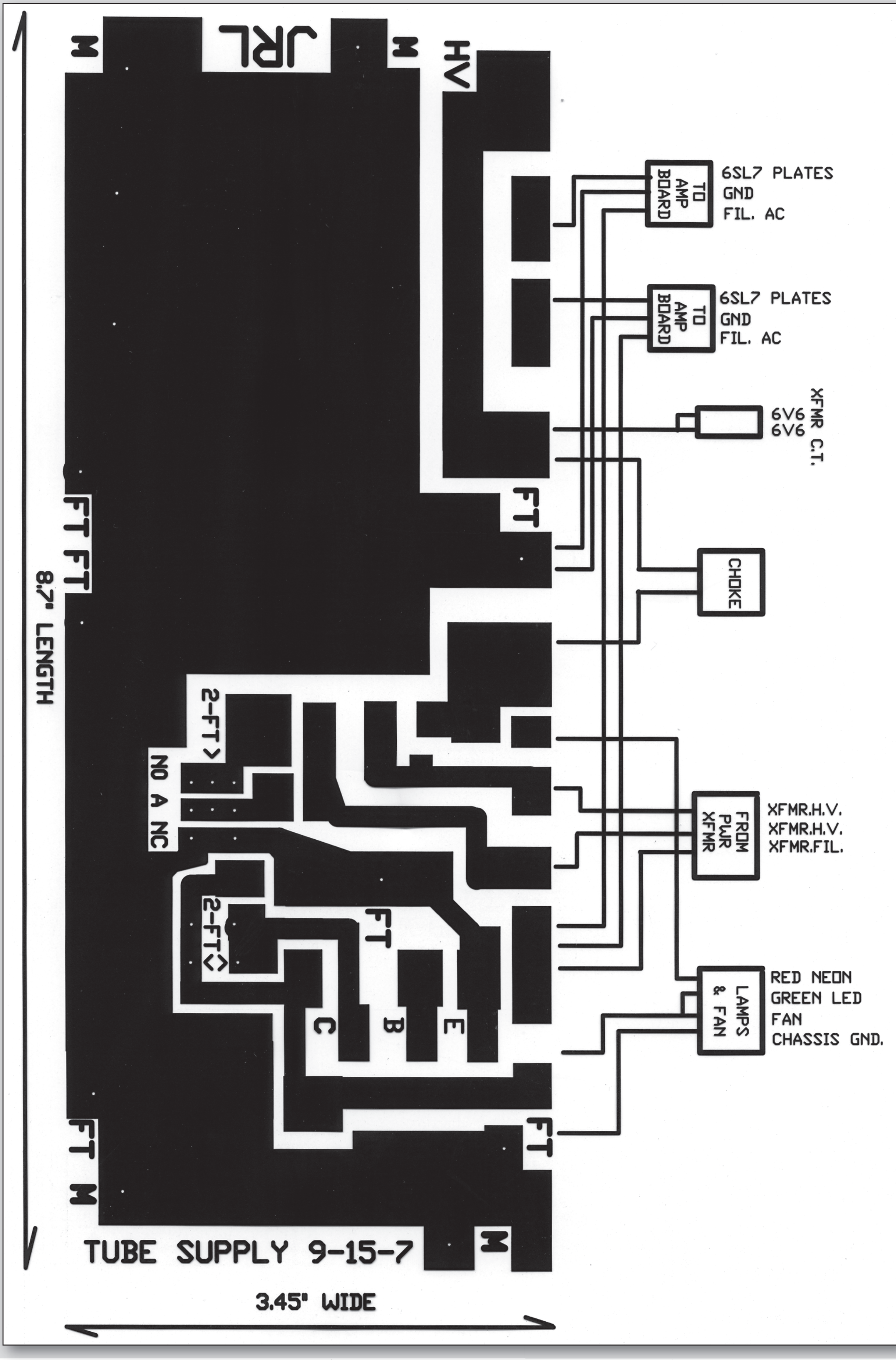


FIGURE E: Power supply bottom layer.

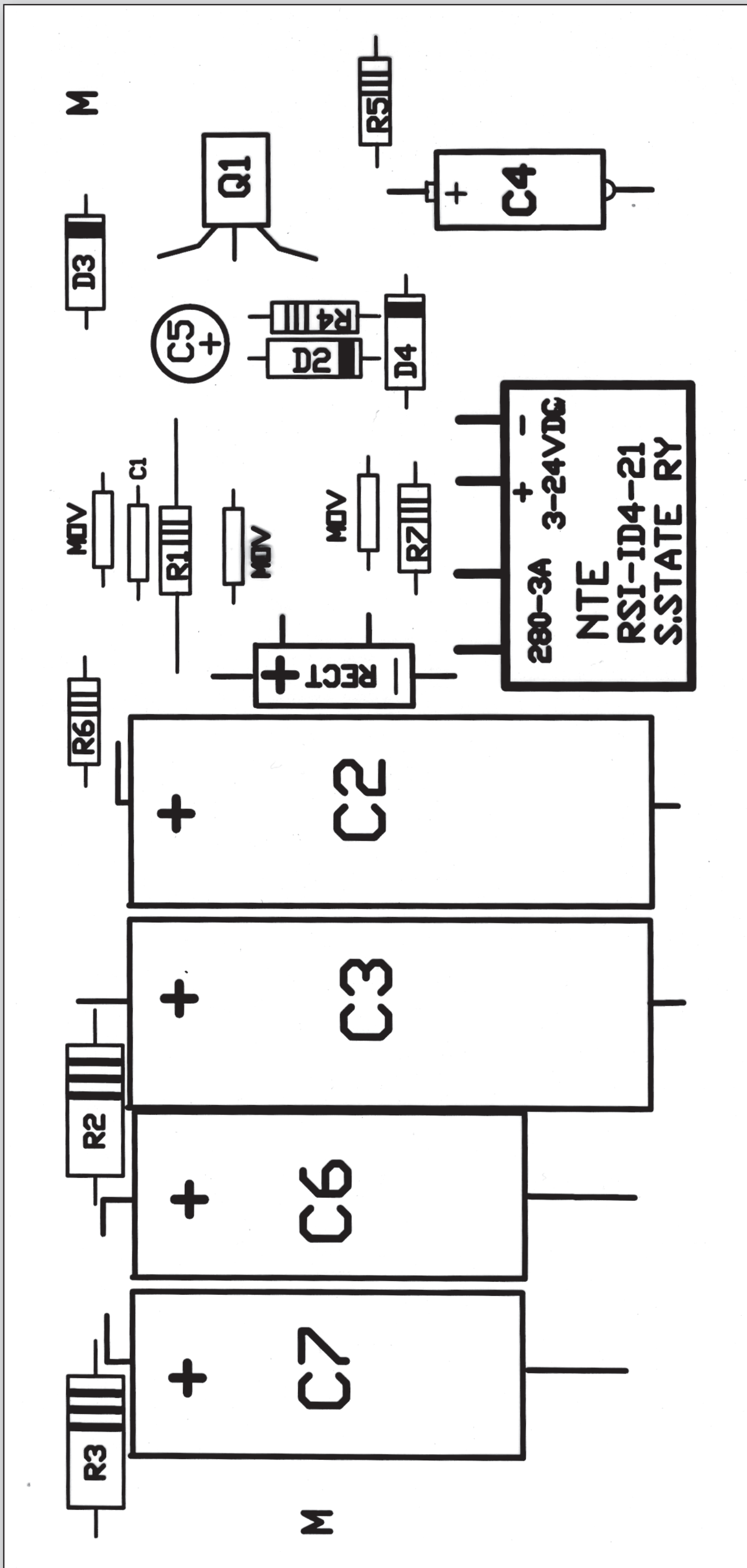


FIGURE F: Power supply bottom overlay.

 By Jack Walton

The Noise (Analysis) Machine

This author reveals a method to conduct accurate noise measurements.

Conventional digital voltmeters (DVM) are incapable of measuring RMS noise beyond a few tens of kilohertz. Even expensive DVMs such as the Hewlett-Packard 3478A, 3456A, or Tektronix DM511 suffer from bandwidth limitations, some of which are "baked in" for RFI avoidance, and others attributable to the deficiencies of the RMS converter. Jim Williams of Linear Technology discussed the shortcomings in a paper, "Performance Verification of Low Noise Low Dropout Regulators[1]."

In addition to the shortcomings of random signal measurement, Dolby, Robinson, and Gundry[2] in "CCIR/ARM Practical Noise Measurement" found that unweighted and A-weighted noise measurements did not correspond well to the human perception of noise in recording. The authors suggest a circuit for a CCIR-ARM weighting filter and measurement regimen that addresses the problem. This paper was written at the dawn of cassette recording, and noise in the vicinity of 6kHz was thought to be under-measured by the systems employed at that time.

What's a DIY audio enthusiast to do? In measuring noise, you could use a preamplifier such as Dennis Colin's low noise measurement amplifier employing the LSK389 dual JFET,[3] or Geller's "JCan" using the BF244A JFET,[4] in conjunction with the RMS detector described here. I have used both the LTC1966 and LTC1968 from Linear Technology and the performance exceeded my expectations, surpassing all of the conventional digital voltmeters I have on hand, and approaches that of the two thermopile "true RMS" meters, a Fluke 8920A and Hewlett-Packard 3403C, which we employ for RF measurement. The CCIR/ARM filter used in conjunction with the RMS detector will allow you to more knowledgeably measure audible noise in your system.

THE CCIR (ITU) ARM CURVE

Dolby's paper includes a complete discussion of the benefits of the CCIR. Very briefly, un-weighted noise measurements are useful in the laboratory when bandwidth is specified, but of very little use in audio. A-weighting, adapted by the National Association of Broadcasters, is based upon measurements of ear-sensitivity (Fletcher-Munson characteristics) and was widely used, but did not correlate well with the observed impression of noise experience. The DIN standard of the time, relying as it did on expensive (at the time) quasi-peak reading measurement, was dismissed owing to limited manufacturing sources and "unappealing" measurements. (In plain English, the DIN standard, if applied to manufactured products of the time, would have resulted in noise figures several decibels worse than advertised.)

The authors recommended the CCIR response curve, one which features a somewhat exaggerated response from 2 to 12kHz, together with an average-responding RMS millivoltmeter. The "zero loss, zero gain" frequency of the CCIR filter was selected by Dolby as 2kHz and is illustrated (with this adjustment) in comparison to the A-weighted curve in **Fig. 1**.

The effects of repetitive impulse noise are understated with an average responding meter, but the authors demonstrate that the differences are close to constant. Thus, the response is compensated by 6dB to take into effect the difference

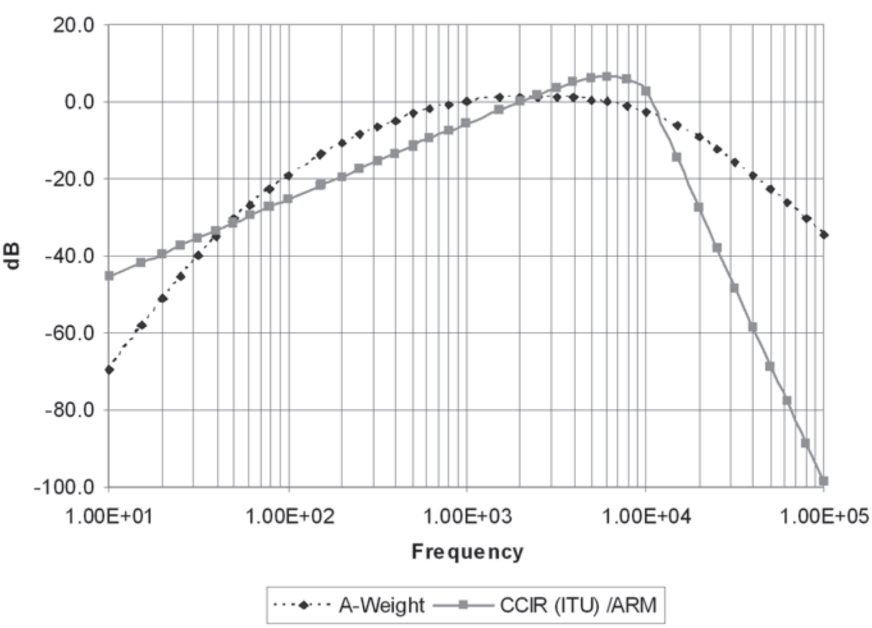


FIGURE 1: Weighted and CCIR filter response.

between measurement regimens. The zero-crossing point was shifted by Dolby from a nominal 1kHz to 2kHz to avoid measurement problems associated with the particular shape of the curve.

A complete description of the noise measurement method is contained in "Measurement of Weighted Noise In Sound Program Circuits, ITU-T Recommendation J.16,[5]" which is available on the ITU website. The paper also contains tolerances at discrete frequencies from 31.5Hz to 31.5kHz. The circuit in **Fig. 2** is used to specify the ITU recommendation J.16.

IMPLEMENTATION

In the Dolby paper the authors showed how to implement the time constants of the filter with a discrete component design consisting of a 7-pole low-pass filter and 1-pole high-pass filter. Simulating their design, I found that the filter was optimized for the 0.0dB shoulder points of the bandpass response.

Today, the filter may be fashioned with operational amplifiers. For the most part, I used the filter values of R&C described in the *J*AES preprint. Not surprisingly, the CCIR filter for my Boonton 1120 Distortion Analyzer used virtually the same configuration, and the schematic I utilized and show is only slightly different from the Dolby and Boonton designs. The Hewlett-Packard (now Agilent) 8903 Signal Analyzer also used a somewhat different arrangement. The HP 8903 CCIR-ARM filter is available on the web[6].

U1A is a first-order low-pass filter with an f3 of 10.6kHz. P1 may be adjusted for 12.2 or 17.8dB gain at a frequency of 6.3kHz. The shoulders of the response curve will then fit close to 0.0dB at 1.0 and 12.5kHz for the ITU Standard or 0.0dB at 2.0 and 10.9kHz for Dolby's implementation. (In this example, I set the shoulders at 1.0 and 12.5kHz.)

The other op amps are 2nd-order Chebyshev low-pass filters with approximately 0.6dB ripple. Feed-forward (R13, R14) was used by Dolby to taper the response from 9.0 to 31.5kHz. The high-pass filter formed by C2, R1, and P2 shapes the response to a bandpass filter conforming to the ITU standard. P2 may be adjusted (as described later in this article) to shift the high-pass response so that symmetry between the 1.0 and 12.5kHz shoulders is attained. The schematic is in **Fig. 3**.

The values of R13 and R14 differ from those of the *J*AES preprint. I found that 392Ω and 95.3kΩ yielded a falloff more consistent with the theoretical values of the LCR filter in the ITU standards. The value of R13 influences the slope of the low-pass filter and can be adjusted to suit your preference. I have used values from 27k to "open." The choice of op amps is not that critical, provided that the gain bandwidth product is sufficient.

RESULTS

The filter performs quite well in comparison to the theoretical values. The performance is illustrated in **Fig. 4**.

The ITU J.16 specification states the tightest tolerance, 0.0dB error at 6.3kHz. The gain of each filter can be adjusted for exactly 12.2dB gain at this point. The gain at 1.0kHz should then be 0.0dB within a tolerance of ±0.5dB, and 0.00dB at 12.5kHz with a tolerance of ±1.2dB. The operational amplifier filter fits well within the specified tolerance bands as illustrated in **Fig. 5**. The tolerance bands specified by the ITU are shown for reference.

I have also provided results for the op

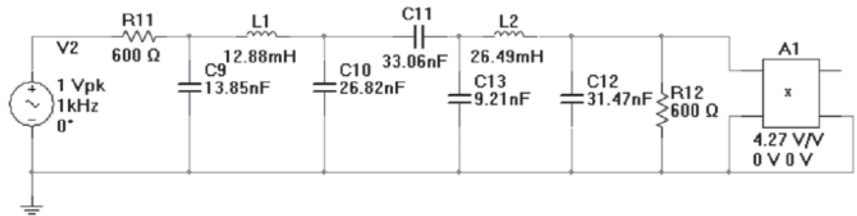


FIGURE 2: Circuit used to specify ITU recommendation J.16.

amp filter without feed-forward (triangular marker). The error in this case is very low for almost the entire curve, but breaks the band at 20.0kHz by about 0.5dB.

The discrete design schematic in the *J*AES preprint is a bit difficult to read, and the slight variance from that of the op amp design may be attributable to errors in reading the circuit diagram from an original that was not well copied. These are simulated results (Dolby Sim, circular marker), not a device that is actually bread-boarded, so reality and SPICE may not necessarily be on the same page. The authors allowed trimming resistors in several locations that would benefit the already good performance.

I also tested the filter with a GenRad 1381 Noise Generator (50kHz bandwidth) and an HP3577 Network Analyzer (**Fig. 6**).

SETUP AND USE

While the setup should be straightforward, you may prefer to use the "snippable" machined IC socket pins for placement of R13 and R14. You can then experiment with values to render an optimal response.

Use of a voltmeter with a dB function is most helpful. In this case, apply a 1.000V 1.0kHz signal to the filter and set the dB "relative" to measure 0.0dB at this frequency. Next, measure the output at 6.3kHz, which should be approximately +12.2dB. If the 6.3kHz output is too low, adjust potentiometer P1 (gain adjust) to bring it to +12.2dB. Reduce the frequency to 1.0kHz and adjust potentiometer P2 to bring the reading at this frequency to 0.0dB.

By adjusting potentiometer P2 (symmetry), gain at 1.0kHz and 12.5kHz can be balanced. When you adjust P2, however, the 6.3kHz gain will also change, so you need to twiddle the little potentiometers until you have a happy *ménage-à-trois* among the 1.0kHz, 6.3kHz, and 12.5kHz settings of 0.0dB, +12.2dB, and 0.0dB. For relative measurements, the filter set up this way correlates very well with the ITU standard. (To match the Dolby CCIR-ARM setup, the 0.0dB crossing points are shifted to 2kHz and 10.9kHz by adjusting the gain of the amplifier.)

If your voltmeter does not have a decibel function, start by setting your function generator for a 1.000V output at 6.3kHz. The output at 1.0kHz and 12.5kHz should be 0.2455V at each shoulder frequency, a level that is -12.2dB from the value at 6.3kHz. You will need to adjust potentiometer P2 to arrive at the blissful settings just mentioned.

The feed-forward network of R13, R14 adjusts the slope of the low-pass filter. Without R13 (infinite resistance) the results were very accurate, except at one data point, 20.0kHz. You can try different resistance values to come up with a filter shape that suits your purpose (the easiest being elimination of R13!).

TRULY MEASURING AN RMS VOLTAGE

One of the reasons for writing this article was my desire to use Linear Technology's LTC1966 or LTC1968 "True RMS Detectors." Even high-quality digital multimeters are woefully inadequate

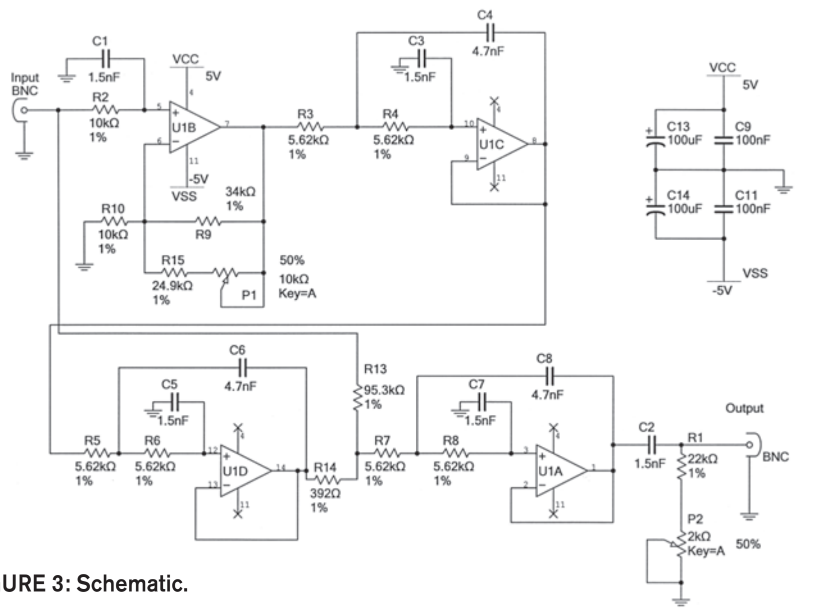


FIGURE 3: Schematic.

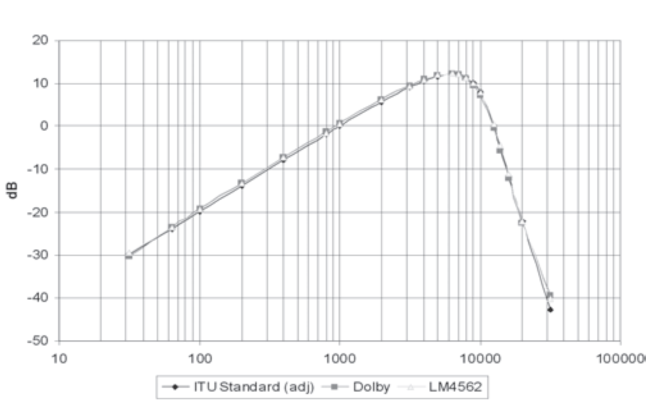


FIGURE 4: CCIR-ITU ARM filters.

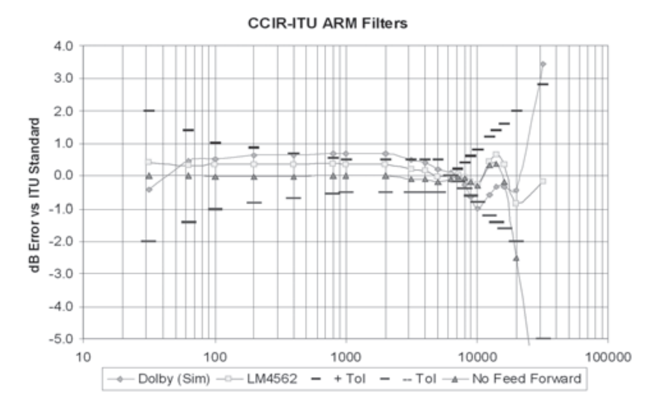


FIGURE 5: Discrete Dolby filter compared with operational amplifier.

when attempting to accurately measure voltages above a few kilohertz or to measure complicated or random voltages. Above some 10s of kilohertz, the performance of many very high-quality voltmeters are compensated to eliminate the influence of radio frequency and electromagnetic interference. **Figure 7** demonstrates the frequency limitations of popular meters, along with the superiority of "truly true" RMS meters such as the Fluke 8920A and Hewlett-Packard 3403C, or the RMS measurement function on the Tektronix TDS3012B oscilloscope.

The "log-antilog" RMS converter, usually an Analog Devices AD536 used in the HP3478A or Boonton 1120, doesn't cut it when it comes to making noise measurements over a few tens of kilohertz. On the other hand, the Fluke 8920A and HP3403C meters, using thermopile true RMS detectors, excel but are very difficult, probably unrepairable if the thermopile is damaged. Jim Williams of Linear Technology points out their measurement superiority in his aforementioned application note, "Performance Verification of Low Noise, Low Dropout Regulators."

Recently, Linear Technology introduced the LTC1966, LTC1967, and LTC1968 true RMS detectors. These replaced the now obsolete LT1088, a thermally responding true RMS detector that was discussed in AN-83. The LTC1966, 7, and 8 use a proprietary (and patented) delta-sigma modulation technique in which the converter's output is equal to $(V_{in})^2/V_{out}$. The output is averaged with a capacitor that forms a low-pass filter.

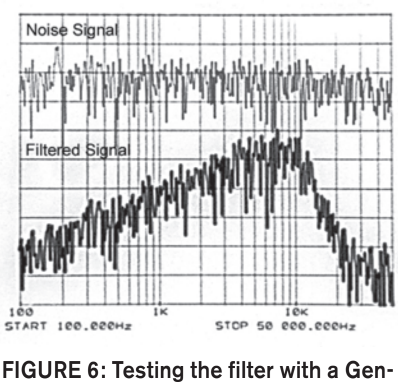


FIGURE 6: Testing the filter with a Genrad 1381 noise generator (50kHz bandwidth) and an HP3577 network analyzer.

The description of the operation of the device is beyond the scope of this article, and I refer you to the product description on Linear's website. The LTC1968's modulator clocks 0.8pF at 2MHz, while the LTC1966 clocks 2.5pF at 100kHz. Linear indicates that the LTC1966 has greater linearity than the LTC1968 but more limited bandwidth. On the other hand, the LTC1966 can use ±5V DC supplies, while unipolar operation of the LTC1968 is recommended.

These devices require a bit of care

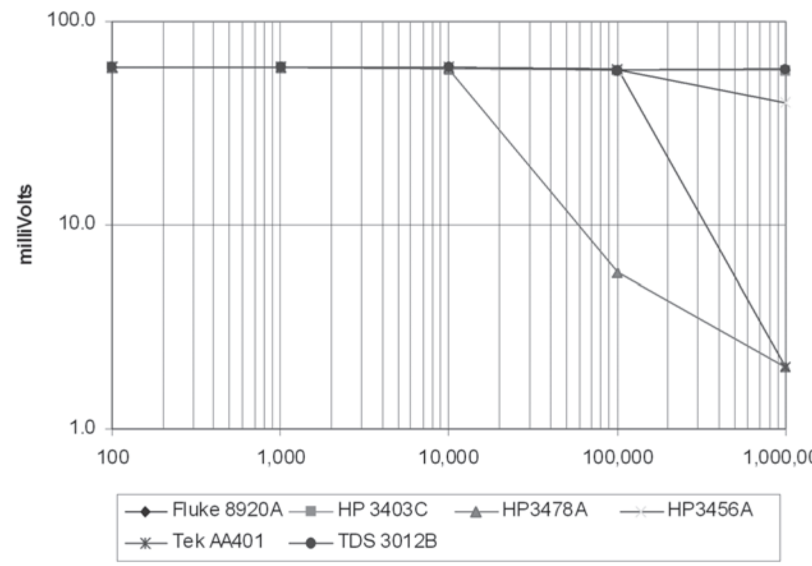


FIGURE 7: "True RMS" meter performance.

and feeding for proper use. First, they are packaged as MSOP8, which makes a printed circuit board a necessity. (You can get around this by purchasing the development board sold by Linear). Second, I found that the best results were obtained when the device input was buffered with a simple non-inverting op amp. Third, a quality averaging capacitor such as a polycarbonate, polypropylene, or polyester type will improve accuracy. WIMA recently introduced its MKS2-XL series, which offers a 10μF 16V DC PET capacitor in a MKT07X9R5 profile, which I use as the averaging capacitor.

Last, ripple is present on the output due to the switching action of the modulator. This is removed with a low-pass filter. I found that the "DC-Accurate" post filter worked best, eliminating ripple and rendering a very easy-to-use DC converter.

I used a Linear Technologies low-power LT6203 dual operational amplifier to serve as input buffer and post filter for the RMS converter. The PCB shows a through-hole DIP pinout. I simply used the LTC6203 on an SOIC-DIP adapter. The schematic for the buffered and filtered RMS converter using the LTC1968 with a single supply (and LT6203 with dual supply) is shown in **Fig. 8**.

RESULTS

In an input range of 100mV to 1000mV, the errors for the LTC1966 average approximately 0.07%. **Figure 9** illustrates the errors for DC- and AC-coupled situations of both the LTC1966 and the AD536.

The data in the LT product folders is formatted in terms of absolute error, not relative error. **Figure 9** certainly confirms the excellent performance of the RMS converter for voltages greater than 100mV. The comparison between devices is on an apples-to-apples basis, using the post filter and averaging capacitor recommended by ADI, but without offset trim.

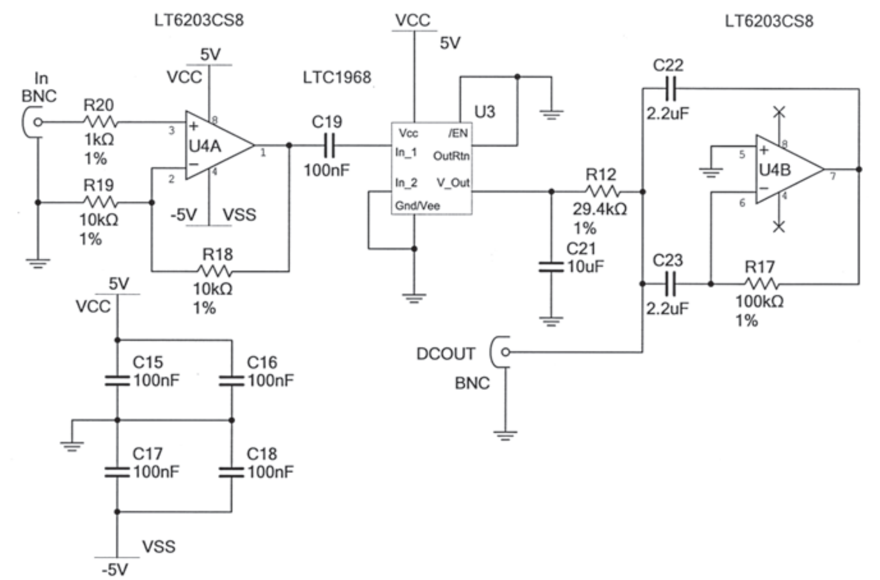


FIGURE 8: Schematic for buffered and filtered RMS converter using the LTC1968 with a single supply (and LT6203 with dual supply).

PARTS LIST

Part Number	Value	Shape
PWR1	1	header, 1 × 3
DCOUT, Input, Output, RMS 1	connector	HDR 1 × 2
C1-3, C5, CV7	1.5nF	capacitor, metal film
C9, C11, 16, 17	100nF	ceramic capacitor
C13, C14	100μF/25V	aluminum electrolytic capacitor
C15	100nF	surface mount ceramic capacitor 0603
C21*	10μF	capacitor, metal film
C22, C23	2.2μ	capacitor, metal film
C4, C6, C8	4.7nF	capacitor, metal film
P1	10k	trimmer potentiometer
P2	2k	trimmer potentiometer
R1	22k 1%	resistor, ¼W
R2, R10, R18, R20	10k 1%	resistor, ¼W
R3-8	5.62k 1%	resistor, ¼W
R9	34k 1%	resistor, ¼W
R11	1k 1%	resistor, ¼W
R12	29.4k 1%	resistor, ¼W
R13	95.3k 1%	resistor, ¼W
R14	392 1%	resistor, ¼W
R15, R16	24.9k 1%	resistor, ¼W
R17	43k 1%	resistor, ¼W
R19	100 1%	resistor, ¼W
U1	LT1014	quad operational amplifier
U3	LTC1968	true RMS detector
U4	LTC6203	dual operational amplifier

***Wima MKS2-XL 10μF 16V DC recommended.**

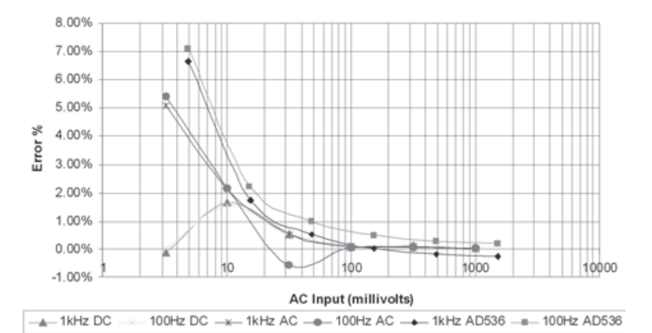


FIGURE 9: LTC1966 and AD536 performance.

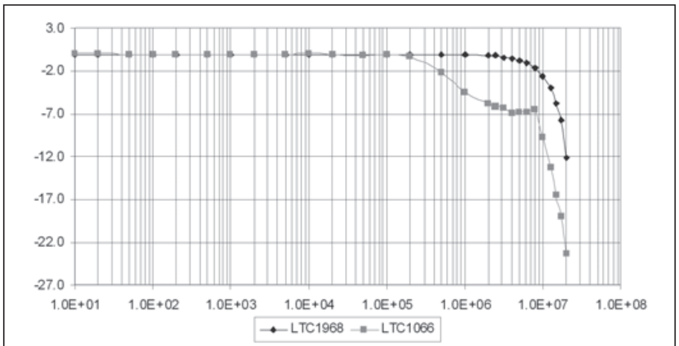


FIGURE 10: LTC 1966 and 1968 bandwidth (measured with DC Accurate filter).

BANDWIDTH

The LTC1968, used on the development board with the DC accurate filter option switched in provides accurate measurement to 1.0MHz, and is down 3dB at 12MHz (**Fig. 10**). The LTC1966 on the filter PCB is a bit more constricted in its bandwidth, but according to Linear, more linear. The peculiarities in the response curve of the LTC1966 may be owing to my own “inartful” design. Either filter chip would be good in the measurement of audio noise, but the LTC1968 would be preferable in a digital voltmeter, or in the gain control circuitry of an oscillator.

A TOUGHER TEST

The Audio Precision System 2022 analyzer provides several flavors of random and pseudo-random noise. **Figure 11** compares the LTC1968 with the Fluke 8920A, HP3478A, and my handheld Fluke 177.

The conclusion is pretty evident. If you choose to measure noise in your system, be aware of the limitations of the RMS detector in your measurement instrument. The LTC1966 and LTC1968 chips from Linear Technology offer great performance at a comparatively low price. Furthermore, the use of the CCIR-ARM filter will help you to accurately measure the noise you may or may not hear!

Test equipment used in this article includes Audio Precision System 2022 analyzer, Boonton 1120 Analyzer, Hewlett-Packard 3336B Synthesized Level Generator, Hewlett-Packard 3577A Network Analyzer, GenRad 1384 Noise Generator, Fluke 8920A True RMS Meter, and Tektronix TDS3012B Oscilloscope.

PRINTED CIRCUIT BOARD

Both circuits can be fashioned on one printed circuit board measuring 4.5 by 2.0″. The top, bottom and silk (stuffing guide) are illustrated in **Fig. 12** and at www.audioXpress.com *aX*

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1. Williams, Jim, “Performance Verification of Low Noise Low Dropout Regulators,” Linear Technology Corporation Application Note 83, March 2000.

2. Dolby, Roy; Robinson, David; Gundry, Kenneth; “CCIR/ARM: A Practical Noise Measurement Method,” Audio Engineering Society Preprint 1353, May 1978.

3. Colin, Dennis, “A Low-Noise Measurement Preamp,” *audioXpress*, April 2007, pp. 26-33.

4. Geller, Joe, “Build the JCAN to Measure Resistor Noise,” *Nuts and Volts*, July 2007, pp. 54-62.

5. The URL for the standards wiki is http://en.wikipedia.org/wiki/Standard:ITU-R_468. The URL for the ITU is www.itu.int.

6. Agilent put the HP8903 service manual on the web at www.agilent.com and inserting 8903 in the search engine.

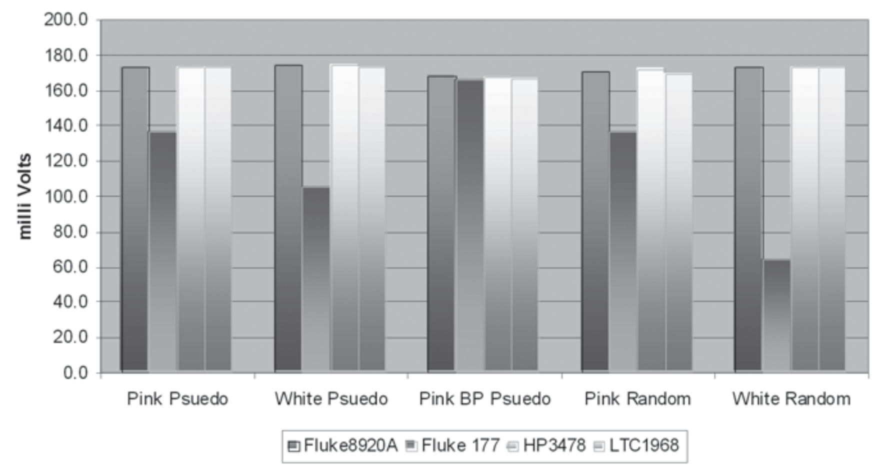

FIGURE 11: True RMS detector comparison.

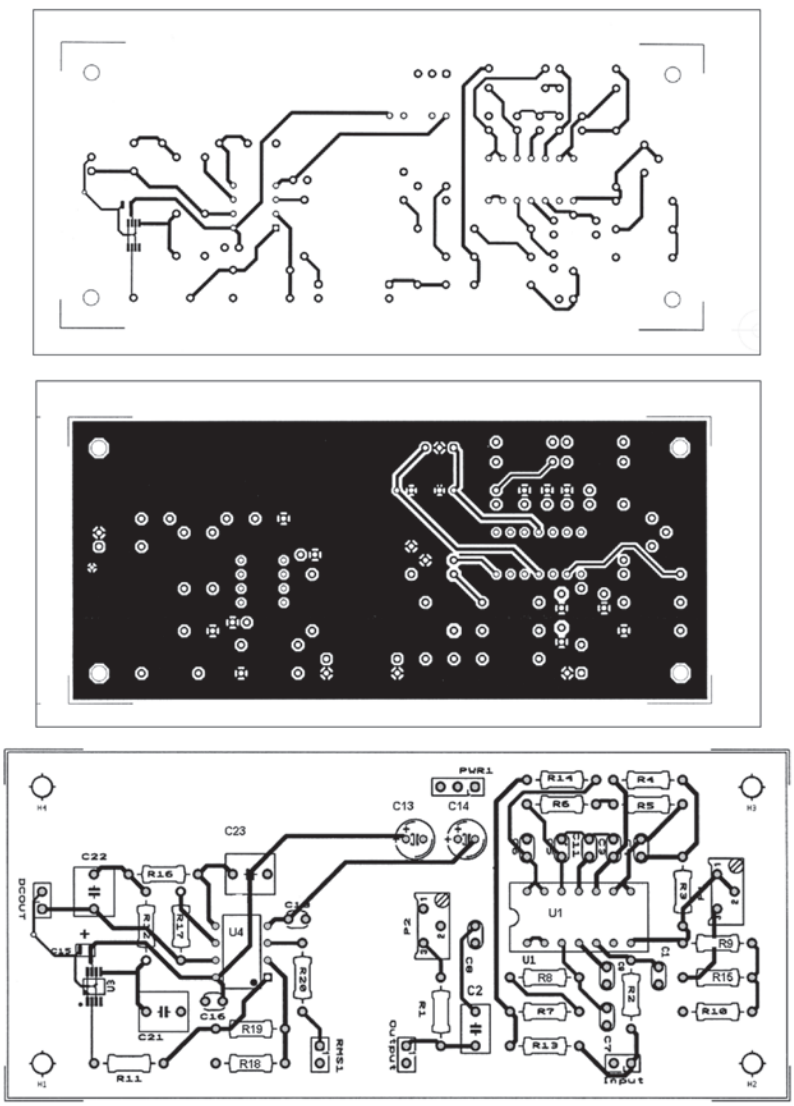

FIGURE 12: Printed circuit boards: top, bottom, and silk (stuffing guide).

By Aren van Waarde

The G4OEP: A Flea-Powered Headphone Amp

This low-power amp design gives new meaning to the word "simple."

Would you believe me if I raved about the audiophile qualities of a car audio chip? I cannot blame you if you didn't. Last spring I constructed a headphone amplifier using a pair of TDA2003 ICs, and this is how it came about.

Since January 2006, I've subscribed to the British magazine *SPRAT*, which, as some of you may know, is an amateur radio quarterly published by the G-QRP club and edited by the Reverend George Dobbs (call sign G3RJV). Its pages are "devoted to low power communication." Some club members have established radio contacts over a distance of 10,000 miles using intelligent software, *very* slow Morse code, and just 0.5mW of power! Others have called their ham friends in another city using a lemon as the only power source.

CIRCUIT FEATURES

Occasionally there are more down-to-earth articles in *SPRAT*. In the summer 2006 issue, Dr. Andrew Smith (call sign G4OEP) wrote "High Performance Headphone Amplifier,"[1] which described a circuit based on a well-known car audio chip, the TDA2003 (**Fig. 1**). This simple circuit comes straight from the datasheet[2], but some component values have been altered in order to reduce the voltage gain (from 100 to 18), and reduce the output power (from 4W to 0.01W).

You can easily reach both goals by increasing the overall feedback ratio, with the additional benefit of a 25-fold reduction in harmonic distortion. Gain is determined by the ratio of R2 over R1 (**Fig. 1**). In the standard application, R2 is 220Ω and R1 is 2.2Ω, resulting in a voltage gain of about 100. In Dr. Smith's circuit, R2 is 2k2 and R1 is 130Ω, so that the gain is lowered to (2200/130)+1 = 18.

High-frequency cutoff (F3) is determined by C5 and R2, according to the formula:

$F3 = 1/(2 * \pi * R2 * C5)$

Thus, when C5 = 2.2nF and R2 = 2k2, F3 is set to 33kHz. Low-frequency cutoff is mainly determined by C6 and the impedance of the headphones.

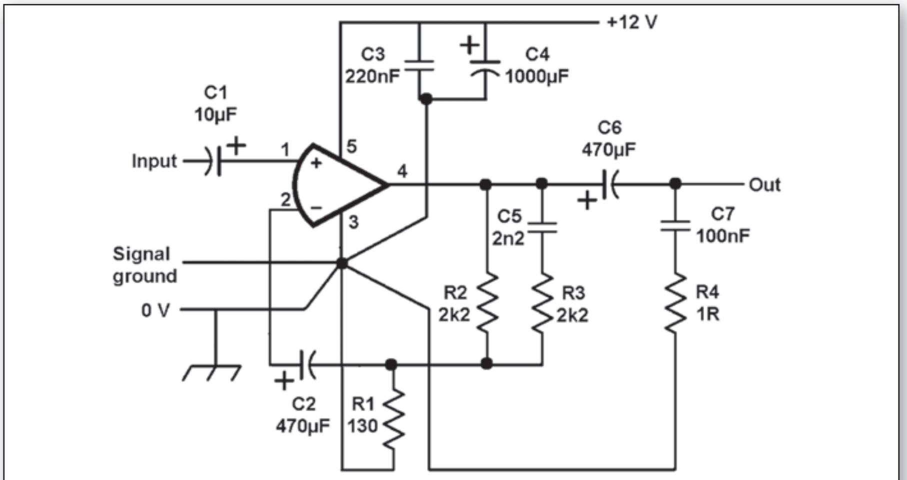

FIGURE 1: Amplifier schematic (one channel). There is a 30k log potentiometer (volume control) at the input (not shown).

TABLE 1A: PARTS LIST (FOR ONE AMPLIFIER CHANNEL)

R1	130Ω 0.25W
R2, R3	2k2 0.25W
R4	1Ω 1W
C1	10µF 100V radial electrolytic
C2, C6	470µF 40V axial electrolytic
C3	0.22µF 100V MKT
C4	1000µF 40V axial electrolytic
C5	2.2nF 100V MKT
C7	0.1µF 100V MKT

IC1 TDA2003 (SGS Thomson). Heatsink not necessary.

NB: There is a discrepancy between the article in SPRAT and the schematic published on the G4OEP website[4]. In the SPRAT article, R3 has a value of 2k2, but in the posting on the web this resistor is 330Ω. The latter value will result in larger bandwith but also in greater risk of instability.

TABLE 1B: POWER SUPPLY (ONE FOR EACH AMPLIFIER CHANNEL)

C8-C11	10nF 400V ceramic (not absolutely necessary)
C12	1000µF 40V (in my prototype 2 × 470µF 40V radial electrolytic, in parallel)
C13, C14	0.1µF foil cap
D1-D4	1N4002 (I used SY330/2, which are 200V 0.5A fast rectifiers from the former GDR)
IC2	L7812CV (SGS Thomson). Heatsink not necessary
S1	Mains switch (dual-pole dual-throw)
Tr1	Transformer 230V: 2 × 12V, PCB type, 10VA minimum

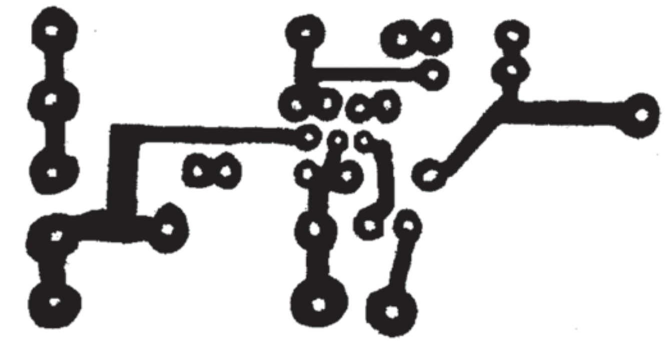

FIGURE 2: Amplifier PCB (solder side, one channel).

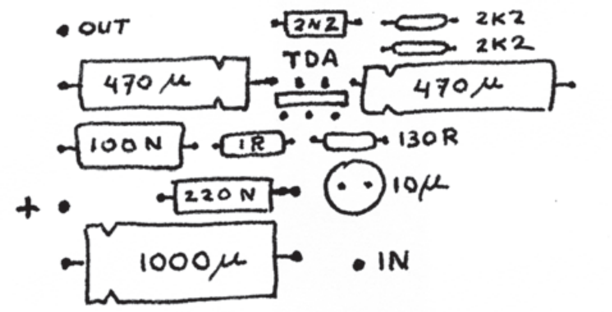

FIGURE 3: Parts placement (one amplifier channel, seen from solder side).

In my prototype, I measured a -3dB low-frequency cutoff frequency of 13Hz (left channel) and 14Hz (right channel), using a 32Ω load. When you use headphones with higher impedance (such as the 85Ω Sennheiser HD25SP of Dr. Smith) the low-frequency cutoff will be shifted to 10Hz (**Table 2**), and even less in the case of AKG or Beyer dynamic.

According to Dr. Smith, "Objectively the performance of this amplifier is flawless, and it also sounds sweet." The circuit "gives excellent performance, equal to the very best commercial designs costing hundreds of pounds[1]." I found this hard to believe, because the TDA2003 is a Class B amplifier, and the proposed circuit makes liberal use of negative feedback. Both aspects seem incompatible with good sound. However, I considered the following:

(a) At the small amount of power required by sensitive headphones, the TDA2003 will operate continuously in Class A.

(b) Well-designed op amp circuits with overall negative feedback can sound good, as Dieter Burmester[3] has proven.

(c) My junkbox contains several TDA2003 ICs that I bought in Germany for only 38 Eurocents apiece (www.pollin.de). For such a small price, you can run the risk and perform some tests! Thus, I decided to give the G4OEP circuit a try.

TABLE 2: AMPLIFIER MEASUREMENTS

(by Dr. A. Smith)
Voltage gain: 18
Nominal power output: 10mW
Maximum output reached at: 200mV input
Harmonic distortion: 0.006% (at 1V peak out in 85Ω)
Spectrum of distortion (1V peak in 85Ω at 1kHz):
Only 2nd and 3rd harmonic visible
2nd at -90dB
3rd < -88dB
Noise: -100dB
Hum: -84dB (dependent on PSU circuit and layout)
Frequency response: 10Hz – 50kHz (-3dB) in 85Ω

(my measurements)
Frequency response: 14Hz - > 30kHz (-3dB) in 32Ω
Quiescent current: about 40mA at + 12V

(the 85Ω load consisted of Sennheiser HD25SP headphones)
(the 32Ω load consisted of Grado SR125 headphones)

APPLICATION

From experience I know that the TDA2003 chip can be a difficult animal—it is prone to oscillation if the circuit wiring is poorly laid out. The *SPRAT* article warns: "To avoid instability problems, a rigorous single-point earthing scheme should be used. The common earth point should be as close to pin 3 as possible[1]."

Following this advice, I designed a simple PCB in which R1 and R4 are very close to pin 3. I ensured that the input and output pins of the circuit were at opposite corners of the board, i.e., at a great distance from each other (**Figs. 2** and **3**). Dimensions of the PCB (3.5 × 2″, or 9 × 5cm) are based on the use of 40V axial electrolytic capacitors from Philips (HP-LL-M), which, in my opinion, have an excellent price-performance ratio. My parts placement is indicated in **Fig. 3**.

With this home-made PCB, TDA2003 chips from SGS-Thomson,

LISTENING TESTS

During the listening tests, I was impressed by the audio quality of the following CDs:

1. Tielman Susato, "Dansereye" (1551). New London Consort. Philip Pickett, conductor. Decca 436-131 2 [Producer: Peter Wadland, engineers: Jonathan Stokes and Neil Hutchinson, recorded 1991, released 1998].

A collection of Renaissance dances performed on a great variety of instruments, including such exotic items as hurdy-gurdy, rebec, cittern, sackbuts, serpent, crumhorn, curtals, shawms, racketts, sordins, rommelpot, and regal. Although originating from the early 16th century (probably even from medieval times), several of these melodies can still be heard in folk music today (listen, e.g., to recordings of the Flemish ensemble Het Kliekske on CBS). My favorites are "Hoboeckendans" (track 16), Pavane La Battaille (i.e., The Fight, track 26), and Entre du Fol (i.e., Entry of the Fool, track 37). The disk is delightful, although occasionally there is some stridence in loud passages, and the performance may be a tad too polished for this repertoire.

2. "Wim van Beek plays the organ of the Dutch Reformed Church in Kantens." Works of Sweelinck, Scheidt, Van den Kerckhoven, Böhm, Buxtehude, Pachelbel, and anonymous 16th century composers. Orgelcommissie Kantens, WBK2001 [CD released in 2001, engineer: Jan Willem van Willigen. Can be ordered via the Internet, www.groningenorgelland.nl/cd.htm Price 10 Euros].

This is a marvelous CD in every respect, capable of converting even die-hard organ haters. The 17th century organ at the village church of Kantens is a beautiful and unique instrument. The music is joyful, easy on the ear, well played, and very well recorded. Until his retirement, Wim van Beek was organ teacher at The Royal Conservatory in The Hague and the Conservatory in Groningen.

3. Jacob van Eyck, "Der Fluyten Lusthof" [The Flute's Garden of Delight]. Played by Erik Bosgraaf on various recorders. With Izhar Elias, baroque guitar, and Inmaculada Muñoz Jiménez, pandereta. Brilliant Classics 93391 [Box of three CDs released in 2007, engineer: Peter Arts, producer: Thiemo Wind] See www.brilliantclassics.com, www.jacobvaneyck.info and www.erikbosgraaf.com.

Jacob van Eyck (1589-1657), the blind city carilloneur of Utrecht, played his recorder on summer evenings in the Janskerkhof. The public, strolling in the churchyard, was overwhelmed by his virtuosic art. The solo variations, preludes, and fantasias (based on British, French, Italian, Dutch, and German folk songs, besides some melodies from the Geneva psalter) were printed during van Eyck's lifetime in "Der Fluyten Lusthof." For this recording, Erik Bosgraaf selected 69 of the 150 published works. He plays this music very well, with great freedom of timing and dynamic expression. The CDs are also well-recorded (in a chapel at Warmond, near the city of Leiden). The accompanying booklet is written by Thiemo Wind, who recently defended a doctoral dissertation about van Eyck. For several weeks, this recording occupied the no. 1 position of the Top 50 of classical music releases on Radio Netherlands. Highly recommended!

4. La Bella Noeva: Les Chants de la Terre. [Madrigals of Giulio Caccini, Alessandro

Grandi, Biagio Marini, Claudio Monteverdi, Giovanni Stefani, and an anonymous Italian composer.] Marco Beasley, tenor. Ensemble Accordone. Guido Morini, organ/harpsichord/artistic direction. Alpha 508 [recorded and released in 2003, engineer Hugues Deschaux, see www.alpha-prod.com].

A technically and musically perfect rendering of early 17th century Italian love songs. [I may be biased because I am a great fan of Marco Beasley and his ensemble]. The music is full of emotional expression. "La Bella Noeva" means "good news," and the different love stories are told in a very lively way. The disk finishes with a madrigal based on the biblical Song of Songs, a short version of the Laudate Dominum (setting of Psalm 150), and an alleluia.

5. G.F.Händel, Gloria (World Premiere Recording), Dixit Dominus. Emma Kirkby, soprano, Royal Academy of Music Baroque Orchestra, Laurence Cummings, conductor (Gloria), Hillevi Martinpelto, soprano, Anne Sofie von Otter, alto, Stockholm Bach Choir, Drottningholm Baroque Ensemble, Anders Öhrwall, conductor. BIS CD1235. [Released in 2001, engineers Ingo Petry and Robert von Bahr].

In 2001, the manuscript of Händel's Gloria was rediscovered in the library of the Royal Academy of Music at London. This CD contains the world premiere recording of this unknown work, besides an excellent performance of the Dixit Dominus, a Latin setting of Psalm 110. The acoustics of Duke's Hall at the Royal Academy of Music (Gloria) are rather dry, those of the Adolf Fredrik Church in Stockholm (Dixit Dominus) are more resonant. The voices of soloists and choir are well recorded, with naturally sounding sibilants. And Händel's music is very beautiful.

6. Friedrich Hartmann Graf, "Out of the Shadow of the Masters." Sonata in e for flute and basso continuo; trio in D for flute, violin, and cello; quartet in G for flute, violin, viola, and cello; quartet no. 2 in D for two flutes, viola, and cello; quintet in Eb. Ensemble Schönbrunn. Globe GLO 5212 [Producer: Paul Janse, engineer: Paul Peter Polak, recorded in 2004, released in 2005, see www.globerecords.nl].

This disk contains what probably are the first recordings of several unpublished works by F.H. Graf, a contemporary of Joseph Haydn. The manuscripts are in the Gjedde Collection at the Royal Danish Library in Copenhagen. They were "rediscovered" by Martin Root, flute player and leader of the Schönbrunn Ensemble. Graf's music is surprisingly diverse: some of his works resemble pieces by Johann Christian Bach or Joseph Haydn, the quintet is more in the style of the later chamber music of Schubert. There are many unusual innovations in Graf's scores, but the outcome is always interesting and pleasing to the ear. The Schönbrunn Ensemble (consisting of eight musicians from Great Britain, Germany, and Holland) plays with great virtuosity and musical perfection, and the recording is excellent. Strongly recommended for anyone who can appreciate music not belonging to the "iron repertoire."

7. F.Mendelssohn-Bartholdy, Paulus (oratorio opus 36). Melanie Diener, soprano, Annette Markert, mezzo-soprano, James Taylor, tenor, Matthias Görne, bass, Choeur de La Chapelle Royale, Collegium Vocale Gent, Orchestre des Champs Elysées, Philippe Herreweghe, conductor. Harmonia Mundi HMC 901584.85 [2 CDs] Live recording, Auditorium Stravinsky, Montreux 1995 [released in 1996].

Although this is a live recording, voices sound natural and the strings are very sweet. Mendelssohn's oratorio (based on the biblical Book of Acts) contains several well-known (Lutheran) hymn melodies. The performances by the four soloists, the double choir, and orchestra are excellent. The album has an informative booklet. Buy this cheap release if you can, you will not be disappointed.

8. Bach after Bach. Famous works of J.S. Bach transcribed by Max Reger for piano 4-hands (Brandenburg Concertos nos. 2 and 6, Orchestral Suite no.3, Toccata and Fugue in d). Wyneke Jordans and Leo van Doeselaar, fortepiano. Challenge Classics CC 72070 [Recorded and released in 1998, producer: Marcel Schopman, engineer: Jorn Mineur].

To most modern listeners, transcriptions of Bach's music in a late Romantic style are anathema. But any doubts you may have are dispelled after the first few notes. Wyneke Jordans and Leo van Doeselaar (a married couple) play with such joy and vitality that you *must* listen to the music. The sound of the Bechstein piano is very well recorded. A truly remarkable production. ■

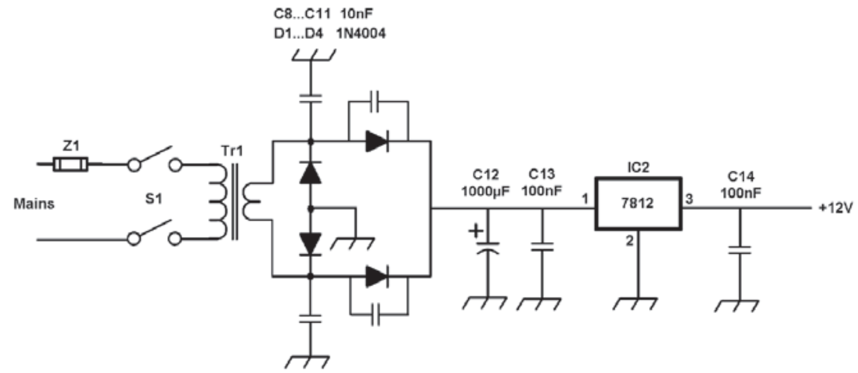

FIGURE 4: Power supply (one channel).

and the wiring scheme of **Photo 1**, there is no instability. Each channel of the prototype has its own power supply (**Fig. 4**). The transformer is shared between the channels, but because it has two independent secondaries, the PSU is a dual-mono construction. The supply is very simple and based on L7812CV voltage regulators[4].

For listening, I mainly used a Grado SR125 (32Ω), occasionally also using a Sennheiser HD465 (60Ω). A Sony Discman and a Marantz CD80 player were the digital audio sources. It turned out that the claims of G4OEP are absolutely correct. The amp is very quiet. There is no audible hum and noise during operation. Take care with the orientation of the power transformer: a 90° rotation can make the difference between slight hum and no hum at all.

Switching transients during power-on or power-off are minor. It sounds both sweet and precise. Differences in recording quality or in the acoustics of different recording venues are immediately evident. Percussion instruments and sudden transients are well-reproduced, indicating good control of the headphone by the amplifier. Sibilants are not artificially emphasized,

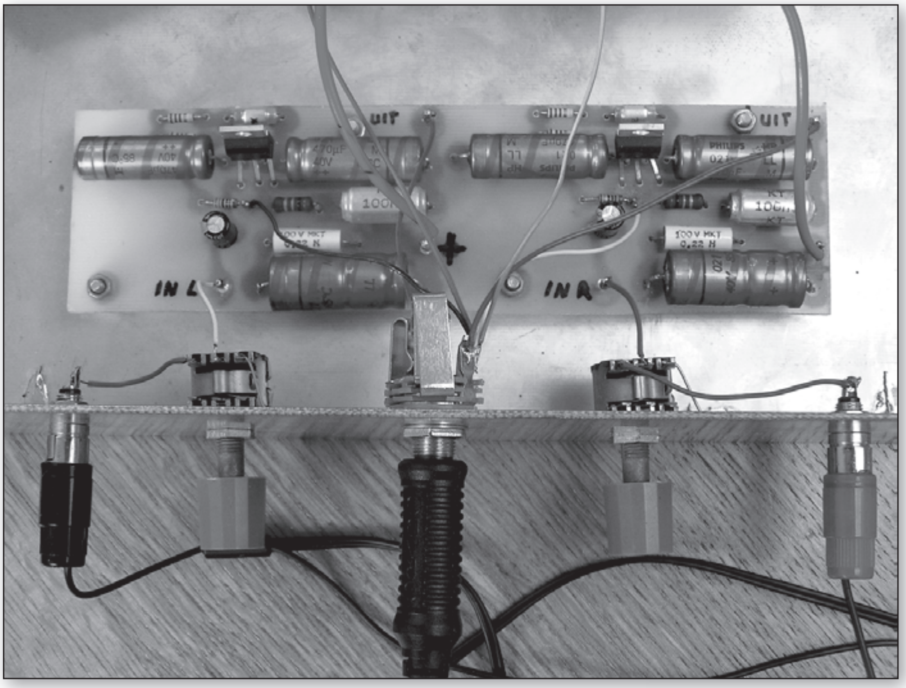

PHOTO 1: Prototype of amplifier built on a "chassis" made of double-sided PCB material. All signal earth and PSU ground wires run to the headphone jack which serves as a star ground.

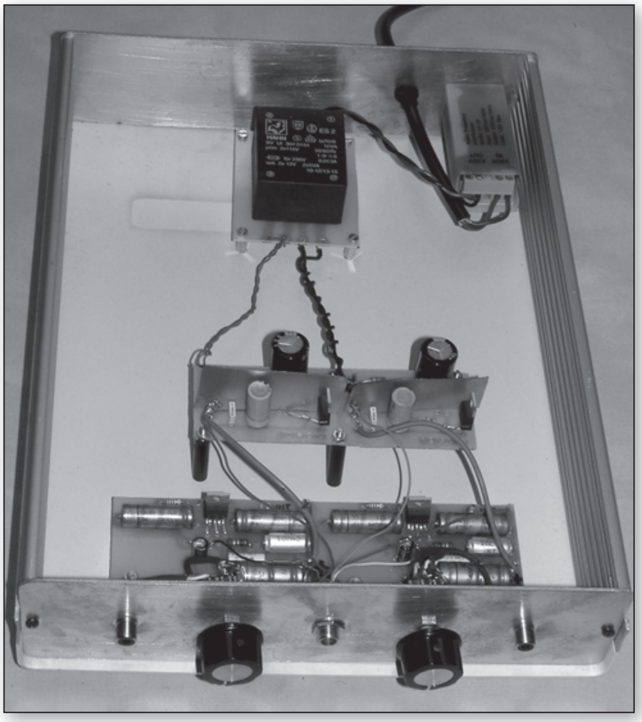

PHOTO 2: Amplifier built into metal cabinet.

and voices sound very natural when the recording is good (unfortunately, many recordings are rather poor in this respect). In conclusion: I love it and will soon place it in a nice-looking cabinet!

This novel design is known as "The G4OEP Headphone Amplifier" after its inventor, but also as "Il Semplice." This Italian expression means "the simple one." I can't think of a headphone amplifier simpler than this! Semplice is also a musical term, meaning "simple, unaffected, without embellishments." The musical presentation of the G4OEP headphone amplifier is like that: very clean and without any evident flaw.

POST SCRIPT

Since the writing of this article, I have placed the headphone amp with power supply in a spacious metal cabinet which was originally intended to house a computer modem (**Photo 2**). I added a mains filter to the 230V AC power supply line. Screening of the amplifier by the metal cabinet and the addition of the filter resulted in a small improvement of the sound. *aX*

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1. Dr. Andrew Smith, "High Performance Headphone Amplifier," *SPRAT* 127:27-28, 2006.
2. SGS-Thomson, TDA2003: 10W Car Radio Audio Amplifier. You can download the datasheet from: http://eu.st.com/stonline/books/pdf/docs/1449.pdf.
3. Dieter Burmester is the owner and chief engineer of the company Burmester Audiosysteme, a German manufacturer of high-end audio equipment, based in Berlin (www.burmester.de). Many of his preamp designs use op-amp circuits.
4. SGS-Thomson, L7800 Series Positive Voltage Regulators. You can download the datasheet from: http://eu.st.com/stonline/books/pdf/docs/2143.pdf.
5. G4OEP website: http://g4oep.atspace.com, see especially the page: http://g4oep.atspace.com/headamp/headamp.htm.

By Ed Simon

Soldering: A Tutorial

As advertised, this soldering station appears too good to be true.

Mike (note: all names given are friends and/or associates of mine) borrowed my copy of Harry F. Olson's book *Acoustical Engineering* 30 years ago and I haven't seen it since. So while working on a recent project, I borrowed Dave's copy. But to be fair, I decided to replace my copy.

A quick web search showed this 1957 book selling for $150 per copy used. In 1991 Jesse got a pristine copy from Mark and, the copyrights having been expired, reprinted it with an updated introduction.

GOING SHOPPING

My web trip revealed that Old Colony Sound Lab had this version in stock for only $69.95! They also had a few other items of interest. So I ordered two other books and one item that struck my interest: a new soldering station from Velleman—the VTSS5V. This was listed as a 50W temperature-controlled soldering station for only $21.99!

I placed my order through the website. The order entry was a bit different from most commerce sites, but quite functional. A few days later I received an e-mail from Old Colony customer service representative Sharon LeClair letting me know the items were shipping the next day.

I received everything well packed within a few more days. The reprinted book still contains the same error as Dave's older copy. Page 211 Equation 6.68 should be DI = 10 * log base 10 (Q)! The soldering system came in an attractive package touting its virtues as a temperature-controlled unit.

BACKGROUND

I first started soldering with a Weller soldering gun at least a few months ago. Shortly thereafter I graduated to a Marksman soldering iron. I own and have used soldering coppers, an acetylene torch, a propane torch, a butane iron, Weller, Unger, American Beauty, and many no-name soldering irons.

The most common soldering station I use today is the Weller WLC100 variable temperature station. This is pretty much just a soldering iron connected to a light dimmer to adjust the power delivered to the iron tip.

I use two different kinds of solder: 63/37 eutectic (or minimum melting temperature) and silver-filled solder. These melt at different temperatures. I also use these in sizes from 0.020″ to 0.062″, depending on the size of the joint.

When I am fixing a low-cost piece of gear, sometimes the pc foil is so thin that it wants to jump off the card at the slightest excuse, so I use a lower temperature and a finer tip. Other times I may be soldering a ground terminal that wants lots of heat and a bigger tip. So I adjust the power to the soldering iron manually with the control knob.

A proper solder joint takes 3 seconds to make. If it takes more than that to cool, then the temperature is set too high. Longer than that to heat the joint, and the iron needs to be hotter.

The light dimmer electronics in the Weller WLC100 does not monitor the tip temperature; it only adjusts the power going to the iron. So when Tim becomes impatient waiting for it to warm up and turns the control to full, the tip warms up faster and gets hotter than it needs to be. If he forgets to set it back to the normal operating range and leaves it on, the tips do not last very long.

Soldering works so well to join wires because the solder has the ability to dissolve a bit of the copper! Works great for the wires, but rather quickly eats up a copper tip. To prevent this, a thin iron plating is placed over the copper core on the better tips. Of course, I buy the "Iron Clad" tips because they last much longer than the older solid copper tips. This makes them a bit harder to tin, but greatly increases their lifespan. To keep the tip clean and the tinning uniform, most of today's soldering stations have a sponge to wipe the tip.

FEATURES

I was interested in trying this new low-cost temperature-controlled soldering system because a controlled soldering station adjusts the tip temperature by using a sensor to keep tabs on how hot the tip becomes. The control circuit would increase the power if the tip were too cold, and lower it if the tip became too hot. Better tip life and the ability to heat up larger joints, all in one!

The new Velleman station was a bit smaller than the Weller. It had the basics—a stand with an adjusting knob, a sponge, a holder for the iron with a guard, and a ground terminal for connecting an ESD mat or strap.

The tip on the new iron uses a screw-on collar to hold in a fairly long-bodied tip, which appeared to be "Iron Clad." The handle had a softish plastic grip on top of the body. That gives a better grip and helps you keep your fingers away from the hot parts.

The system seemed well made, so I set it up. I first noticed that the sponge is set into a well in the top of the base. I keep a plastic bottle of water with my solder station to wet the sponges. I was not quite comfortable with the idea of filling the sponge well, for fear that water would drip down the front of the soldering station. I compromised by filling the well half full of water and then lowering the sponge into the well. If this was not quite enough water, you can always remove the sponge and add more water.

I find most of my assembly time is spent putting parts in place; soldering is just a small step. So I usually place the iron to my left even though I solder with my right hand. Thus it is out of the way until I need it. The cord provided for the iron was just barely long enough to do this. This system is clearly intended to be used by a right-handed person on his/her right side.

Another possible problem is the small base, which tends to tip over on the left side if you are not careful with how the power cords are arranged. I also managed to not put the iron straight back into the holder and started melting the protective plastic surrounding the metal.

The collar nut used to hold the tip in place loosens up with use, as is typical of every soldering iron I have used. Unfortunately, when you tighten it the entire barrel rotates. This worries me because I do not want the power cord to twist when I tighten the collar. This iron requires two tools to tighten the collar.

HEATING THE IRON

The temperature adjustment knob is calibrated by a color scale yellow to orange to red. This seems quite intuitive and practical, but does require some feel to get to the temperature you find works well for your work.

I used this station on several projects. The tip that came with the unit seemed to be a reasonable size for most of the work. It would be too big for some of the fine pitched lead systems used today, but if you are soldering those items you will most likely use a smaller tip or better equipment.

A better soldering system often uses a low voltage isolated heating element. This system relies on the high temperature insulation and the safety ground wire to prevent damage to sensitive components.

The big advantage of a temperature-controlled iron is that the sensor in the tip is used in a feedback circuit to keep the tip at a constant temperature. My standard Weller WLC100 variable temperature system does not do this. That is why Tim starts it off at the highest setting. A controlled iron would turn on at full heat and then reduce the power to the iron tip automatically as it heats up. The control knob would set the temperature for the size of work and type of soldering you are doing. If you place a well-controlled tip on a larger surface, more power is delivered to keep the tip hot. Let the iron sit and less power goes out, the tips last longer, and less energy is wasted.

The Velleman VTSS5V did not seem to heat up as fast as I expected for a 50W iron. I tried turning up the heat when I first turned it on to see whether it would heat faster. It did! That was not right for a temperature-controlled iron. I opened up the base and there were only three wires going from the control card to the iron. There were two wires for power and one for ground. That's two wires short for a feedback sensor! The unit is not a temperature-controlled system, but just a variable temperature system. That explains the low price.

Old Colony also carries a spare tip for this system. . . only one size. Because this is not a high-end tool, I do not think that this should be a problem for most users, because Velleman does list other tips as available.

Despite its low cost, the poor water well, the bad stand, the short iron cord, and not quite living up to its description make this a *not* recommended product. If you want to save money, try a simple 20 or 25W iron, which will do just about everything this system does for even less money! *aX*

VENDORS

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www.borbelyaudio.com
In Japan:
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Ad Index

ADVERTISER	PAGE
ACO Pacific Inc	31
Ask Jan First	21
Audience	41
Audio Amateur Corp	
Old Colony Sound Lab	
Aluminum Enclosures	42
Bendix Red Bank Tubes	38
Audio Transformers	52
AudioKarma	23
Audiomatica SRL	15
Avel Lindberg	43
E-Speakers	7
ETI – Eichman Technologies	CV2
Guerilla Audio/Moon Dog Records	40
Hammond Manufacturing	3
Jantzen Audio Denmark	8
KAB Electro-Acustics	43
Linear Integrated Systems	51
Madisound Speakers	29
Midgard Audio AS	5
Mouser Electronics	25
Mundorf EB GmbH	11
Parts Connexion	13
Parts Express Int’l., Inc.	CV4
RAAL Advanced Loudspeakers	33
Selectronic	44
Sencore	19
Solen, Inc.	49
Tang Band Industries Co.,Ltd.	50
Tektron-Italia	37
Test Equipment Depot	39
Triad Magnetics	CV3
WBT-USA/ Kimber Kable	53
CLASSIFIEDS	
All Electronics	45
Audio Classics Ltd.	45
Billington Export Ltd.	45
Borbely Audio	45
ENG Vista, Inc.	45
TDL Technology	45

By David J. Weinberg

AES Multi-Tracks in New York 2007, Part 2

More highlights from the 123rd AES Convention in New York City

TUTORIALS

"Here Comes the Sun—Are You Listening?" William Moylan (Coordinator of Sound Recording Technology, University of Massachusetts Lowell; www.UML.edu) believes that "hearing requires learning what to listen for, then developing skill in the process of listening." Through listening to and analyzing *Here Comes the Sun* and other Beatles productions, this tutorial illustrated key ways to "enhance listening skills and learn from those who create the albums we love." I was never sure whether Moylan was saying that music creators intentionally place the instruments with respect to each other and the sound stage, or merely create musical effects that as a byproduct can be interpreted as instrument location and motion. A critical note is the musicians' limitations due to the available number of channels and recording/mixing technology. He asked the audience to decide whether one instrument was louder than a different instrument—a question flawed by not specifying measured or perceived loudness.

"Tinnitus: Just Another Buzz Word?" Dr. Neil Cherian (neurologist specializing in vestibular and balance disorders, Center for Headache and Pain, Cleveland Clinic; www.ClevelandClinic.org) explained that "tinnitus is a common yet poorly understood disorder where sounds are perceived in the absence of an external source. He touched on the "pertinent anatomy and physiology, audiologic parameters, current research, guidelines for identifying high risk behaviors, and how to determine that you have a problem." Tinnitus—an affliction, not a disease—is estimated to affect more than 50 million in the US, of which about 12 million seek medical help. There are many symptoms, with a range of severity, no definitive test, and no correlation to an audiogram. Neither is there a clearly identifiable cause, but some observational coincidences. There is also no specific treatment, and all are off-label, meaning that a medicine used for other purposes *might* have the beneficial side effect of reducing tinnitus' severity.

"High Definition Perception—Psychoacoustics, Physiology, and the Numbers Race." Poppy Crum (psychophysicist, Johns Hopkins University and Johns Hopkins Hospital; www.HopkinsHospital.org) acknowledged that "improvements in digital technology have led to higher fidelity recordings by some definitions, but they have also generated much controversy. Sampling rates and bit depths well beyond the limits of conventionally accepted human perception can be used, but are they necessary?" Crum discussed psychophysical and physiological research on ultrasonic hearing, noting that you don't directly hear ultrasonics in the same way you hear audio frequencies, but you do sense some ultrasonic stimuli. Whether that simultaneously affects hearing in the audible range is unproven. She gave a tutorial on “nonstandard mechanisms of hearing transduction across species as an ecological approach to hearing development.”

Many attendees brought preconceived notions about her topic, with some expecting her to support consumer digital audio at higher than 16-bit/44.1ksps as audibly better than the Red-Book CD standard. Her only claim was that higher-resolution audio recording is beneficial for original recordings and post production due to the degrading effects of extensive processing. She said nothing about the possibly audible difference between the CD standard and higher-resolution recordings. Crum did take issue with the conventional approach to analyzing listening test data, saying that with respect to the September 2007 AES *Journal* paper by E. Brad Meyer and David R. Moran (“Audibility of a CD-Standard A/D/A Loop Inserted into High-Resolution Audio Playback”) and many other papers on listening tests, she doesn’t find the data analysis technique sufficiently powerful to reach conclusions about near-threshold audibility choices. She suggested that signal detection theory, a data analysis approach used in psychophysical experiments, might be more powerful, yielding more reliably accurate conclusions regarding the audibility, or inaudibility, of near-threshold alternatives.

“Loudness Metering Techniques and Standards for Broadcasting.” Steve Lyman (Dolby Labs) chaired a panel consisting of Thomas Lund (TC Electronic; www.TCElectronic.com), Andres Mason (R&D, BBC; www.BBC.co.uk/rd/), and Gilbert Soulodre (Communications Research; www.CRC.ca/en). They discussed the two ITU-R standards, BS.1770 (“Algorithms to measure audio program loudness and true-peak audio level”) and BS.1771 (“Requirements for loudness and true-peak indicating meters”), and how the meters should be used.

“Architectural Acoustics.” Tony Hoover (McKay Conant Hoover Inc.) emphasized architectural acoustics consulting, which is the art of applying the science of sound in a timely, practical, and easily-understood manner. Three distinct areas were covered: sound isolation (airborne and structure-borne), HVAC noise control, and surface treatments (absorption, reflection, and diffusion), plus electroacoustics. Several claims: it takes about 200ms for humans to determine pitch; near-field is closer than the size of the source (earphones are the only near-field monitors); so-called near-field monitors should be called direct-field monitors; for line-array speakers, the level drops about 3dB per distance-doubling until the distance equals the array length, after which the level transitions to dropping at 6dB per distance-doubling.

THE RICHARD C. HEYSER MEMORIAL LECTURE

“This series is an endowment for lectures by eminent individuals with outstanding reputations in audio engineering and related fields. . . It honors the memory of Richard Heyser, a scientist at the Jet Propulsion Laboratory who was awarded nine patents in audio and communication techniques and was widely known for his ability to clearly present new and complex technical ideas. Heyser was also an AES governor and AES Silver Medal recipient.”

“Listening to Music in Spaces.” Few are as worthy of such an honor as Leo L. Beranek, whose book *Acoustics* (in its current edition; ISBN 978-0883184943) is still *the* reference text more than a half century after initial printing. In his early 90s, Beranek is still active in the field, having just completed his 13th book (to be released in March) and recently been acoustical consultant for four concert halls, one opera house, and two drama theaters (I’m almost breathless recounting even these recent accomplishments!).

Beranek addressed “apparent source width, listener envelopment, lateral fraction, inter-aural cross-correlation coefficient, reverberation time, early decay time, initial time-delay gap, strength (and various substrengths including early and late relative levels, and early and late lateral relative levels), perceived bass, texture, just noticeable differences, and instrumentation.” These parameters are being met in halls of different architectures with varying success. He touched on the relation of these parameters to living-room listening venues. To highlight Beranek’s devotion to research, he included ideas from an exceptionally obscure paper he had encountered less than a month before the presentation. Some of the topics were from papers he has recently published, but the overall impact of so much valuable information about concert hall acoustics in one address cannot be overstated.

WORKSHOPS

“Concert Halls and Their Acoustics” was hosted by John F. Allen (High Performance Stereo; www.HPS4000.com), and included panelists Leo L. Beranek, Chris Blair (acoustician; www.Akustiks.com), Barry Blesser (www.Blesser.net), Larry Kirkegaard (architectural acoustics; www.Kirkegaard.com), Jonathan McPhee (Music Director, Boston Ballet; www.BostonBallet.org), and Alan Valentine (Nashville Symphony president; www.NashvilleSymphony.com). These luminaries covered a wide range of issues, including Kirkegaard’s work acoustically refurbishing London’s Royal Festival Hall. The highlight was discussion of the new Schermerhorn Symphony Center in Nashville, Tenn.—including the Laura Turner concert hall that is earning accolades as being in some ways acoustically better than Boston’s Symphony Hall.

Valentine gave a detailed explanation of how the concert hall committee raised interest and financial backing from the city fathers and wealthy local investors, of the committee’s insistence that world-class acoustics be the primary focus, and about the overwhelmingly strong economic benefit that has come to the neighborhood and Nashville in general. The committee visited and studied halls in Europe and the US, with the design inspired by the great halls of the world—a basic shoebox shape of proper size and height, with sound-diffusive walls and ceiling. Allen, who has attended concerts in Symphony Hall for many years, said the Schermerhorn Hall, with about 1860 comfortable seats versus Symphony Hall’s ~2600, has better bass (not an easy accomplishment) and higher sound level.

LUNCHTIME KEYNOTES

"Beethoven's Deafness." Charles J. Limb (neurologist, otolaryngology, Johns Hopkins Hospital) and Patrick J. Donnelly (computer music research, Sound and Music Perception laboratory, Johns Hopkins University) noted that Beethoven is one of the three best-known deaf people (with Helen Keller and Thomas Edison). "Despite recent advances in medical understanding of hearing loss, the cause of Beethoven's hearing loss remains a mystery. [Limb and Donnelly] reviewed Beethoven's musical compositions from the perspective of a musician with progressive hearing loss, and in the context of medical diagnoses based on modern otologic principles. [They also discussed] recent data, including studies of Beethoven's hair, as part of this investigative study of one of the great mysteries of classical music."

TECHNICAL COMMITTEE MEETINGS

I attended a few meetings, and am not permitted to report on the details. However, it is clear that the participants are motivated to try to make standards and practices better, and coordinate their activities with working groups in other standards organizations such as SMPTE and ITU. I recommend that AES members consider joining those committees whose themes intersect with their interests. Only through involvement can you improve the status quo.

EXHIBITS

ConEQ. In an acoustically treated room, Real Sound Lab (www.RealSoundLab.com) demonstrated its CONEQ™ automatic speaker/room equalization. The system is flawed in that it presumes and requires a radiation pattern that exhibits uniform frequency response versus horizontal angle, a condition very few speakers can meet. When I raised the issue, they seemed unconcerned.

MISCELLANEOUS

High-resolution audio and ABX listening tests. During a non-session discussion, someone raised the question of whether testing increases the listener's stress level (is the listener trying too hard?), altering his hearing sensitivity to differences. Another asked if you listen differently during tests than when you listen, even critically, for pleasure. No conclusions were reached, but the thoughts percolate.

"Geoff Emerick/Sgt. Pepper." Commemorating the 40th anniversary of *Sgt. Pepper's Lonely Hearts Club Band*, Beatles engineer Geoff Emerick explained with examples and personal observances how the album was created.

There were many sessions I could not attend:

- **"Surround Sound: The Beginning, 1925-1940"** (Robert Auld; www.AuldWorks.com), **"Surround Sound: Quadraphonic Sound in the 1970s—Recording Classical Music"** (Auld, moderator; panelist Max Wilcox [www.ClassicsAbroad.org/max-wilcox.htm]).
- **"The Art of Space: Audio Engineers as Aural Architects"** was the subject of a lunchtime keynote address by Barry Blesser (AES Fellow; author of *Spaces Speak, Are You Listening? Experiencing Aural Architecture*; ISBN 978-0-262-02605-5; MIT Press; $40). "Space and sound are inexorably linked because space changes our experience of sound, and sound changes our experience of space."
- **Audio Education Forum.** Mark Parsons (Parsons Audio, Wellesley, Mass.; www.PAudio.com) moderated this panel discussion on preparation for audio-related employment among Jim Anderson (AES convention chair; Chair, Clive Davis Department of Recorded Music, NYU; www.JimAndersonSound.com), Eddy B. Brixen (EBB-consult; www.EBB-Consult.com), Alex Case (University of Massachusetts Lowell), Paul Foeckler (Digidesign; www.Digidesign.com), and Dave Moulton (Sausalito Audio Works; www.SausalitoAudio.com).
- **"Audio Delivery Specification."** (AES preprint 7182) Thomas Lund (TC Electronics; www.TCElectronic.com) addressed the lack of correlation between conventional metering (peak, average, and so on) and loudness. He offered the ITU-R BS.1770 standard as "a coherent alternative to peak level fixation." He presented novel ways of visualizing short-term loudness and loudness history.
- **"Surround 5.1 Mixing Dos and Don'ts"** was a workshop chaired by George Massenburg (www.Massenburg.com) with Frank Filipetti (Right Track Recording; www.RightTrackRecording.com), Crispin Murray (technology development manager, Metropolis; www.Metropolis-group.co.uk), Ronald Prent (Galaxy Studios; www.Galaxy.be), and Jeff Wolpert (www.JeffWolpert.com).
- **Graham Blyth Organ Concert.** As at each AES New York convention, Graham Blyth (co-founder and technical director, Soundcraft; www.Soundcraft.com) gave an organ recital, sponsored by the Soundcraft Studer Group, on the chapel organ and the sanctuary organ in Brick Presbyterian Church. The concert featured an all-British first half (Parry, Stanford, Matthias) plus the complete Widor *Symphony #5*.
- The convention offered a slate of technical tours of college music departments, studios, theaters, and stations.

The PA systems were fairly well adjusted throughout the convention. Intelligibility was quite good in almost all cases, and adequate in those few that were less successfully amplified. The only needed improvement is to lower the lights in the session rooms, especially on the screens, to a level that is high enough to take notes (and keep people from nodding off) while making it much easier to read the projected slides.

CONCLUSION

AES executives, and convention chair Jim Anderson in particular, should be proud of producing such an exceptional convention with a breadth and depth of presentation topics that were uniformly interesting and informative.

Preprints of the technical papers, and a CD-ROM of the entire set, are available at www.AES.org. MP3-on-CD recordings of the sessions and panels are available from Conference Copy Inc (www.ConferenceMediaGroup.com; 800-575-0580). aX

WALL WARTS—PROS AND CONS

I'd like to add another reason to Paul Stamler's list ("In Praise of Wall Warts," Jan. '08 *aX*, p. 24) of why to use wall warts—hum.

Physical separation of the power transformer from low-level audio circuitry is the best safeguard from magnetic pickup from the power transformer. I've battled this problem on many audio projects, once even resorting to mu-metal shielding of the internally mounted power transformer to solve a pesky hum problem. Using a wall wart is the easier solution. This is especially important for low-level circuits most susceptible to hum, such as preamps.

Yet another benefit of wall warts is ease of replacement. Power transformers are prone to failure by power line overvoltage conditions or by power supply circuitry faults (shorted regulator, dried up lytics, and so on). It's much easier to debug and replace a wall wart than an internal transformer.

The circuit in Fig. 1 is a voltage doubler, but is properly used as a single-ended doubler, not a dual-output doubler. By using the pin labeled V- as ground, the pin labeled V+ is double the voltage of a half-wave rectifier circuit. Alternately, by using the common node of the two filter caps as ground, V+ and V- provide a dual-output (but not doubled) supply. Here's a good link on this: www.play-hookey.com/ac_theory/ps_v_multipliers.html.

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Much as I agree with Paul Stamler's enthusiasm for the good points of wall warts, I believe that he failed to mention a few negatives.

First, wall warts have a habit of multiplying. As anyone who works with computer equipment lately has probably noticed, all of a sudden you have *lots* of wall warts and a shortage of properly spaced outlets to plug them into. There are neat products that help a lot, including wide-spaced outlet strips and "squid plugs"—you can guess how they work—but you still end up with a little area dedicated to wall warts that looks sort of like the "tank farm" section of an oil refinery. If your preamp used two wall warts, it would probably occupy most of the area on a standard narrow six-outlet power strip. Avoiding this is one of the benefits of equipment with *internal* power supplies and simple (and usually standard and interchangeable) line cords.

Second, there can be important safety issues with certain configurations. Most wall warts deliver their voltage on one of those small concentric plugs, which have quite tight spacing (especially internally), and a thin and thinly insulated wire. If, for example, you float that filament supply on +160V to center it, that +160V is now being applied to that thin wire and small connecter—and the internal wiring of the wart itself—all of which were only intended to see 16 or 24V. Even if the transformer *in* the wart is rated safe for that breakdown voltage, I doubt that the interconnect wire they use is rated for it. Even if it is, I would not trust the wire typically used in terms of mechanical safety; I wouldn't want dangerous voltage on a wire that could be so easily damaged if you step on it or set something on it.

Third, and last, most wall warts are not "green." Most of them limit their safe current by virtue of having relatively high winding impedance and low efficiency—if shorted they simply don't draw enough power to burn. This is one of the reasons they are so very safe. Unfortunately, the same considerations lead to a transformer that tends to draw a lot of idle current and have low efficiency when operating (some as low as 50%).

Because the wall wart is always connected, even when the equipment it powers is switched off, a large collection of them will tend to draw quite a bit of extra power. Admittedly, it's probably not significant in terms of your electric bill, but those with a green inclination might object. You should also note that some wall warts do not appreciate having their ventilation blocked by being buried under carpet or stray clutter; even some without vents require free air flow around the case for cooling, so some care may be required in that direction as well.

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CROSS-REFERENCES

I recently read an article by Charles Hansen which he wrote last year. In it he stated that it was difficult to cross-reference Heathkit and other old part numbers. ECG has a master replacement guide for semiconductors that does a great job of this. He mentioned an MPSU45. That's an ECG272, for instance. It has many Heath numbers cross-referenced to ECG numbers. The copy that I have was printed in 1989. I am assuming that ECG is still in business, of course.

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Charles Hansen responds:

Mr. Davis is correct, but note that ECG was purchased by NTE a number of years ago (www.nteinc.com). I use the NTE cross-reference all the time. I also have the 1992 SK Cross-Reference book (from Thomson Consumer Electronics), but SK is now TCE/RCA and to my knowledge they have yet to provide an updated cross-reference.

There is also a nice cross-reference for Tektronix and HP test equipment semiconductors at www.sphere.bc.ca/test/tekequiv.html, www.sphere.bc.ca/test/hpparts.html.

Many of the semiconductor manufacturers provide cross-references to their competitors' part numbers. For instance, Texas Instruments provides their cross-reference at http://focus.ti.com/logic/docs/crossreference.tsp?sectionId=451&familyId=1.

MIKE PERFORMANCE

In George Danavaras' article on the Panasonic Mike Cartridge (*aX* 12/07, p. 28), there are many errors and misrepresentations, and at least some clarifications as to the measurement techniques used.

First, a few comments about Bruel & Kjaer microphones. In general, their performance far exceeds their specifications. The microphone used as a reference microphone (standard on the Type 2230 SLM pictured) is a Type 4155—a model that is no longer available. More important, this microphone is not primarily designed for electroacoustic work, but instead is designed for general product and environmental noise measurements. For loudspeaker testing, a 4191 mike (comparable to our older Type 4133) should be used because of its higher cutoff frequency and reduced distortion. Also, the 2230 body will have an effect on the microphone responses; normally only a microphone and its preamplifier body would be used near the tested loudspeaker driver.

I fully understand that the basis for many of the tests made were based on live comparative frequency response measurements using the 4155 microphone on the 2230 sound meter and the DUT microphone taped to its side. Still, the effects of the 2230 and the adjacent microphone will have an effect that could change—especially at higher frequencies—by a slight shift in position or orientation. Ideally, these measurements should be made with a superposition technique—not taping the microphones side-by-side.

Figure 1 seems to show the response of the tweeter, as measured with the mikes, not the response of the microphones, as described. I know this because the B&K mike response is much flatter than this. This should be made clear in the measurement.

Similarly, the distortion measurement parameters are the distortion values of the drivers and perhaps the Panasonic microphones. It does a disservice to our company to promote those measurements as those of our microphone. The correct technique to evaluate microphone distortion is to use two drivers individually driven by separate frequencies that are not harmonically related. Then, you look at the difference frequency values and measure the spectral parameters not present at the two driven frequencies and their harmonics. This allows you to ignore the loudspeaker nonlinearity.

The author commented that an anechoic environment is required for distortion measurements. This is, in fact, an error. While frequency response measurements can easily be affected significantly by the room acoustics, the near field distortion measurements are practically immune to variation, as long as the room is not a highly reverberant room, or by the unlucky chance the specific frequency chosen is at the room standing wave modes.

Again, in Fig. 5, the author presents a frequency response graph that is of the loudspeaker—not the microphones—yet is described as "response of the microphone."

A comment about the Panasonic microphones: While cheap, they are pressure response microphones and exhibit the rising high frequency response shown in Figs. 3 and 6. This rise in response is not really due to the microphone, but is due to the sound pressure buildup of high frequencies as the wavelength of sound approaches the diameter of the microphone. The B&K microphone is designed to have a high frequency rolloff in its pressure response to compensate for this pressure buildup.

CONTRIBUTORS

Bjørn Kolbrek ("Horn Theory: An Introduction, Part 1," p. 6) is an electronics engineer in Norway, and works as an R&D engineer for Scana Mar-El designing remote control systems for ships. His hobbies include building horn loudspeakers, amplifiers, and amateur radio gear. For the last four and a half years, he has studied horn loudspeaker technology—both theoretical—and practical—with Thomas Dunker.

A specialist in a practical approach to electronics, **John Laughlin** ("Amp6," p. 18) has spent 25 years in the electronic industry, and 25 years as a college instructor. He has a first-class radiotelephone license, and is a member of the Brazos Valley Amateur Radio Club. He has written for several publications, including *Popular Electronics*, *Ham Radio Magazine*, and *QEX*. Besides electronics design, he enjoys making custom billiard cues, sailing, and he has walked across the Grand Canyon three times.

Jack Walton ("The Noise (Analysis) Machine," p. 30) was a chemistry major in college (Fordham in NYC), doing much work in physics where he routinely broke equipment, particularly the diamond saw blades used to cut copper-gold alloys. This was at the dawn of the super-conducting era. He received his first-hand experience with vacuum tubes in instrumentation building low-noise amplifiers for the solid-state lab, repairing instrumentation, and playing with liquid nitrogen. He was also very active in ham radio at the time, building VHF tube equipment.

He decided to pursue an MBA at the University of Chicago and worked in finance at several investment banks before he started his own investment management firm. He is deeply involved in the specialty chemicals business at the moment.

He has moved to sand-state and does much work looking at noise and interference issues with audio equipment. He collects Hallicrafters ham radio equipment, sings tenor in choir and has three grown sons. His wife is a molecular biologist, who has variously worked on cancer stem cells and opioid receptors.

Aren van Waarde ("The G40EP: A Flea-Powered Headphone Amp," p. 36) is a Dutch biochemist working in the field of medical imaging (positron emission tomography.) He has worked as a Ph.D. student and a postdoc at several universities (including Leiden and Yale Universities) before accepting tenure at the University of Groningen. His passion for audio started on his ninth birthday when his parents gave him a Philips kit. Most of his current audio equipment (loudspeakers, radios, tuners, pre- and power amps, both tube and solid-state) is homemade.

Ed Simon ("Soldering: A Tutorial," p. 42) received his B.S.E.E. at Carnegie-Mellon University. He has installed over 500 sound systems at venues including Jacob's Field, Cleveland, Ohio; MCI Center, Washington D.C.; Museum of Modern Art Restaurants, New York; The Coliseum, Nashville, Tenn.; The Forum, Los Angeles; Fisher Cats Stadium, Manchester, N.H.

David J. Weinberg ("AES Multi-Tracks in New York 2007, Pt. 2," p. 46) is an engineering consultant and technical journalist on audio, video, and film technology. He provides audio and home theater engineering consultation and professional on-location digital audio recording services to companies, radio stations, and individuals. He brings to his work an MSEE, a First Class Radiotelephone license, and over 40 years of continued study and active involvement in the audio, video, and computer industries. He is Vice Chairman of the Audio Engineering Society's DC section, a manager in the Society of Motion Picture and Television Engineers' DC section, and membership officer of the Boston Audio Society (www.BostonAudioSociety.org).

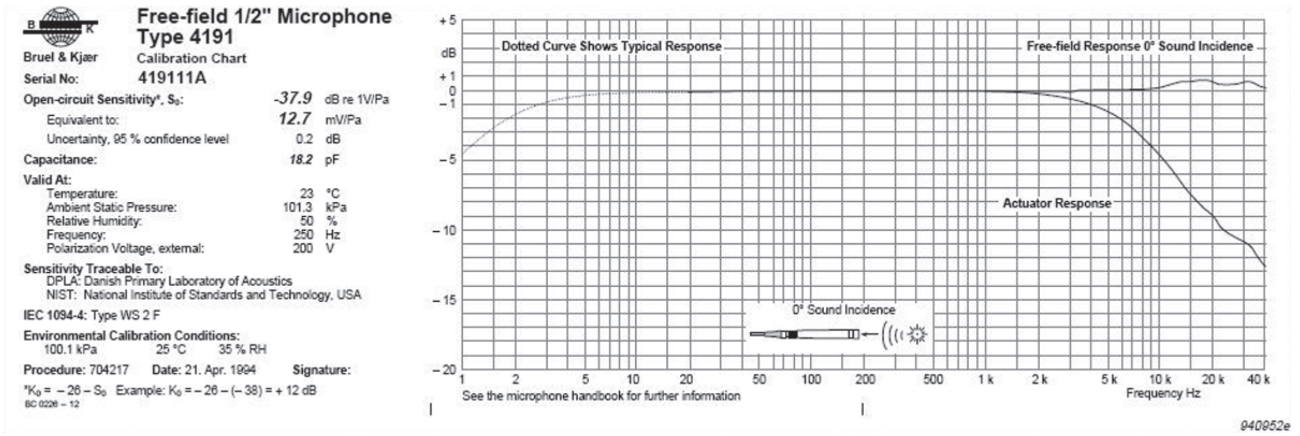

FIGURE 1: Calibration chart supplied with a free-field ½″ microphone type 4191

Figure 1 shows both the pressure (actuator) and free field response on the curves for the 4191 microphone mentioned earlier.

The Panasonic microphones response is severely affected by temperature and humidity variations. This is not too much of a problem in loudspeaker testing that is usually performed indoors and with a controlled environment. Of course, a B&K microphone cannot be affected by external variables, as it is most commonly used outdoors or on factory floors with no control of the environment.

The phase response measurements (Figs. 4 and 7) have me puzzled. Though you did not show the response of the B&K mike, I assume, then, that you are showing the comparative phase of the Panasonic mikes relative to the B&K mike. Because all these have a diaphragm resonance at somewhere around 15kHz, the actual phase variation relative to their low frequency will be at –90° at that resonance. The key is to have the resonance well damped—and all these show a well-damped resonance. This relatively low resonance point is another reason to not use these microphones for high frequency analysis of tweeters. At B&K we recommend 1/4 or 1/8″ diameter microphones with diaphragm resonance frequencies of at least 70kHz for this application.

Also, I would like to mention that to purchase one of our high-quality microphones or sound meters such as our Type 2250, you could spend several thousands of dollars, but we make these products available for rental at prices that would be attractive to an audio club or individual audio enthusiast.

Jim Weir
Sound Level Meter and
Environmental Products Manager
Bruel and Kjaer North America

George Danavaras responds:
I would like to thank Mr. Weir for his letter and the opportunity to make some additional clarifications concerning my article. I agree with Mr. Weir that the 4191 mike would be more accurate for loudspeaker testing than the 4155 mike I used for my tests. I would be happy if I could use the 4191 mike, but I had access only to the 4155 microphone and the 2230 Sound Level meter (SLM).

Concerning the testing method, as I described in the article, I used an acoustic source and I measured the amplitude and the phase response of the source first with the B&K mike and then with the Panasonic mikes. The purpose was to compare the measurements of the acoustic source that were taken with the B&K mike with the measurements of the same acoustic source that were taken with the Panasonic mikes. Then, considering as reference the response taken with the B&K microphone, the response of the Panasonic microphones could be extracted.

The same method applies also for the distortion measurements. I measured the distortion of the acoustic source first with the B&K mike and then with the Panasonic mikes. The distortion figures that are given in Tables 1, 2, and 3 of the article are the distortion of the acoustic source as measured by the B&K mike and the Panasonic mikes.

I think that Mr. Weir misunderstood the meaning of Fig. 1. As I describe in the article, not only does the body of the 2230 SLM have an effect on the microphone response, but also the placement of the Panasonic mike close to the body of the B&K mike also has an effect.

This effect is shown in Fig. 1 of the article. In this figure two amplitude responses of the same acoustic source are shown. Both responses were taken with the B&K mike. One was taken with the B&K alone and the other with the Panasonic mike on the B&K body. The differences in the two responses describe the effect of the placement of the Panasonic mike on the body of the B&K mike.

Due to this effect and some other reasons described in the article, I conclude that the accuracy of the testing method was about ±1dB. Anyway, I believe that most of the misunderstanding comes from the incorrect description of Figs. 1, 2, and 5 and Tables 1, 2, and 3 which should be modified as follows:

Figure 1: Two frequency responses of the same acoustic source are shown. One was taken with the B&K mike alone and the other was taken again with the B&K mike having the Panasonic mike attached on the body. The difference in the responses is due to the proximity of the Panasonic microphone.

Figure 2: The low frequency response of the acoustic source as measured by the B&K mike and the standard Panasonic mikes.

Figure 5: The low frequency response of the acoustic source as measured by the B&K mike and the modified Panasonic mikes.

Table 1: Distortion of the acoustic source at 115 dBSPL peak as measured by the B&K mike and the standard Panasonics.

Table 2: Distortion of the acoustic source at 125 dBSPL peak as measured by the B&K mike and the Linkwitz modified Panasonics.

Table 3: Distortion of the acoustic source at 130 dBSPL peak as measured by the B&K mike and the Linkwitz modified Panasonics.

Mr. Weir notes that I commented that an anechoic enviroment is required for distortion measurements and considers this an error. What I mentioned in the article is that because I don't have an anechoic chamber available, the only solution is the near-field measurement. I don't think that this is an error. Additionally, I don't have any measurements about how the Panasonic mike response is affected due to temperature or humidity. All my measurements were taken indoors at a room temperature about 24° C and humidity 55%.

Finally, I would like to add that the Bruel & Kjaer Company is a world leading manufacturer and its products are the industry reference. If there was any misunderstanding or omission in the article concerning the characteristics of the B&K mike, this was not my intention. *aX*