Acoustic High-Pass, Low-Pass, and Band-Stop Filters

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I. Introduction

Many interesting problems in acoustics involve the propagation of sound in ducts. Furthermore, many applications of acoustics to ventilation / exhaust systems in buildings and automobiles use mufflers and acoustic filters to reduce the level of noise propagating down a duct or radiating from the end of a duct. In this laboratory exercise you will explore the behavior of acoustic waves in a duct with changes in cross-sectional area, side branches, and resonators. In the process you will observe the behavior of acoustic low-pass, high-pass, and band-stop filters as they are used in a duct system.

II. The Theory of Acoustic Transmission Lines

A rather complete theoretical development of acoustic waveguides, transmission lines, and filters may be found in Chapters 9 & 10 of *Fundamentals of Acoustics*, 3rd Ed., Kinsler, Frey, Coppens, and Sanders, (J. Wiley & Sons, 1982).

A. Waveguides and transmission lines

A waveguide is a structure which forces wave propagation along a path parallel to its longest dimension. Acoustic wavequides are structures with constant cross-sectional area and shape. Simple examples of such structures include hoses, tubes, and pipes, referred to hereafter as *ducts*. If a duct is excited by a pressure disturbance with a wavelength larger than twice the duct's largest cross-sectional dimension, then only plane waves will propagate down the duct. For a circular duct containing air at room temperature, the highest frequency at which only plane waves will propagate is given by $f = 100/a$ where a
is the radius of the duct cross-section. Once plane waves are generated inside the duct, they will propagate down the duct, even if the duct has bends or turns in it. A propagating plane wave may encounter a change in the acoustic impedance of the duct when the duct (i) opens into free space, (ii) is connected to another section of duct with a different cross-section, (iii) branches off into two ducts, or (iv) is terminated in some other way. This impedance change causes partial reflection and partial transmission of the incident plane waves.

Assume that a duct of cross-sectional area S and length L is driven by a piston at $x=0$. The pipe is terminated at $x=L$ by an acoustic impedance $\mathbf{Z}_L$. The input acoustic impedance as seen by the driver (looking into the duct at $x=0$) may be written in terms of the terminal impedance,

$$\mathbf{Z}_0 = \frac{\rho c}{S}\left[\frac{\mathbf{Z}_L + i\frac{\rho c}{S}\tan(kL)}{\frac{\rho c}{S} + i\mathbf{Z}_L\tan(kL)}\right] \tag{1}$$

Equation (1) is called a transmission line equation; similar equations are used for electromagnetic waveguides (transmission lines), as well as for comparing the input mechanical impedance to the termination impedance for transverse waves on a string or longitudinal waves in a beam.

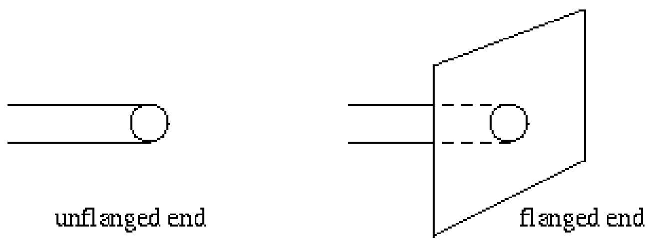
B. Duct driven at *x=0* and open at *x=L*

Let the duct be driven by a rigid piston at $x=0$ and open-ended at $x=L$. At first guess one might think that the termination impedance for an open end would be $\mathbf{Z}_L = 0$, which would reduce Eq. (1) to $\mathbf{Z}_0 = (\rho c/S)\, i \tan(kL)$ resulting in resonance frequencies occurring at $f_n = nc/2L$, for $n=1,2,3,\ldots$ This is the assumption made in most elementary physics books; it is not, however, correct. The boundary condition is not zero at the open end, because the open end of the duct radiates sound into the surrounding medium. The proper value for the terminating impedance is then $\mathbf{Z}_L = \mathbf{Z}_r$ where $\mathbf{Z}_r$

is the radiation impedance of the open end of the pipe. The radiation impedance is complex; the real part (radiation resistance) represents the energy radiated away from the open end in the form of sound waves, and the imaginary part (radiation reactance) represents the mass loading of the air just outside the open end. For *unflanged* and *flanged* open ends, the radiation impedance is

$$\mathbf{Z_r} = \frac{\rho c}{S}\left[\frac{1}{4}(ka)^2 + i\,0.61\,ka\right], \qquad \text{unflanged;} \qquad (2a)$$

$$\mathbf{Z_r} = \frac{\rho c}{S}\left[\frac{1}{2}(ka)^2 + i\,\frac{8}{3\pi}\,ka\right], \qquad \text{flanged.} \qquad (2b)$$



The input acoustic impedance for an unflanged, open-ended duct may be obtained by substituting Eq. (2) into the transmission equation line equation Eq. (1). Resonance occurs when the input impedance becomes a minimum; or when $1/\mathbf{Z}_0$

becomes a maximum. Figure 1 shows the input acoustic impedance calculated from Eqs. (1) and (2a) for a duct of length $L=1.1$ m and radius $a= 2.0$ cm. The first two resonances occur at approximately $f_1 = 170$ Hz and $f_2=340$ Hz.

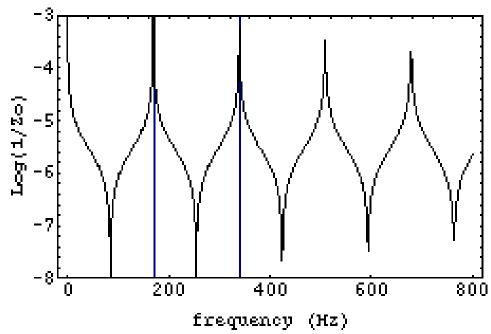


Figure 1: Input admittance (inverse of input impedance) for an unflanged, open-ended duct of length *L=1.01* m and radius *a= 2.0* cm.

The resonance frequencies for an unflanged, open duct may be approximated by

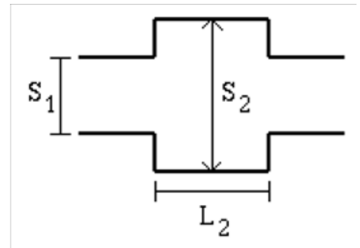
$$f_n = \frac{nc}{2(L + 0.61a)} \qquad (3)$$

where $n=1,2,3,\ldots, c$

is the speed of sound (343 m/s for air), and the length of the duct includes an "end correction" for the open end.

III. Acoustic Filters

A. Low-pass filters



An acoustic low-pass filter may be constructed by inserting an expansion chamber in the duct. An expansion chamber serves as a simple model of a muffler, and also has applications in architectural acoustics (the plenum chamber in a building's HVAC system is an expansion chamber). Keeping track of all incident and reflected waves from both junctions, one can derive the sound power transmission coefficient as

$$T_\Pi \approx 4\left[4\cos^2(kL_2) + \left(\frac{S_2}{S_1} + \frac{S_1}{S_2}\right)^2 \sin^2(kL_2)\right]^{-1} \quad (5)$$

Figure 3(a) shows this coefficient for an expansion chamber with dimensions similar to what you will use in this lab. Notice that there are frequencies at which all incident power passes right through the filter, and there are other frequencies where a minimum of power is allowed to pass by. Frequencies for complete transmission correspond to standing waves being set up within the muffler chamber.

In a low frequency limit (*kl << 1*) the expansion chamber may be treated as a side branch of acoustic compliance $C = V/ p c^2$ where the volume $V = S_2 L_2$. In this low frequency limit, the side branch approximation of the sound power transmission coefficient in Eq. (5) becomes

$$T_\Pi \approx \left[1 + \left(\frac{1}{2}\frac{S_2}{S_1} kL_2\right)^2\right]^{-1} \quad (6)$$

This low frequency approximation is shown in Figure 3(b). The expansion chamber appears to pass low frequencies, and block high frequencies. Thus, it is called a "low-pass" filter.

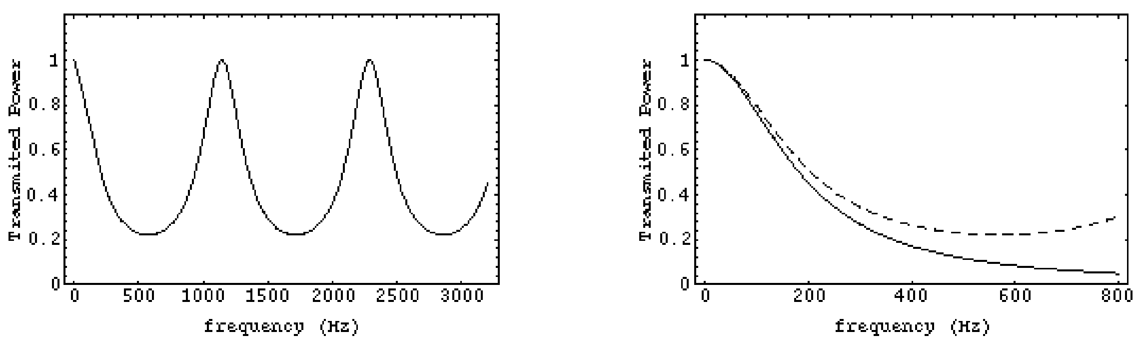
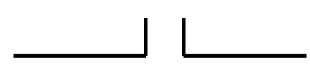


Figure 3: (a) Sound power transmission coefficient for an expansion chamber; (b) low frequency approximation acting as a low-pass filter.

B. High-pass filters



An acoustic high-pass filter may be constructed simply by inserting a "T" junction, or a short side branch into the duct. If both the radius and the length of the side branch are smaller than a wavelengths of the plane waves in the duct then the acoustic impedance of the side branch opening becomes

$$\mathbf{Z}_b = \rho c \frac{k^2}{4\pi} + i \frac{\rho(+1.5a)\omega}{\pi a^2}$$

The first term represents the radiation of sound from the side branch opening, and the second term represents mass loading of the fluid at the side branch opening. For low frequencies the sound power transmission coefficient (which describes how much sound energy makes it past the filter) may be derived as,

$$T_\Pi \approx \left[1 + \left(\frac{\pi a^2}{2S(L+1.5a)}\right)^2\right]^{-1} \tag{4}$$

The transmitted power is shown in Figure 2 for a side branch you may encounter in this lab. This acoustic filter blocks low frequencies and passes high frequencies, thus it is called a "high-pass" filter. It is important to realize that low frequency energy is not radiated out of the side branch, but is reflected back towards the source.

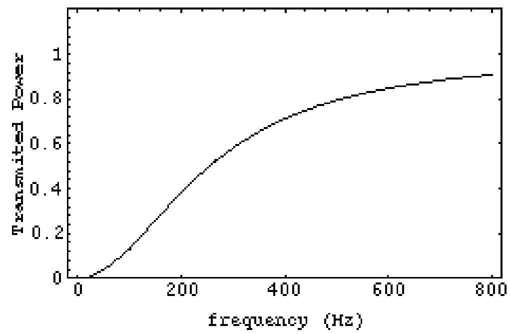
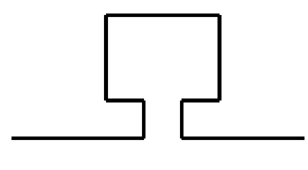


Figure 2: Sound power transmission coefficient for a side branch in the low frequency approximation of a high-pass filter.

C. Band-stop filters



If a cavity is attached to the side branch, as shown at right, then the side branch has both mass (inertia) and compliance. Such an acoustic system is called a Helmholtz resonator --- it behaves very much like a simple mass-spring system (have you ever blown into a pop bottle to make a sound?). This resonator has a neck with radius *a* and area S_b, an effective neck length of $L_{\text{eff}} = L + 1.7a$, and a cavity volume *V*. This cavity resonates at a frequency

$$\omega_0 = c\sqrt{\frac{S_b}{L_{\text{eff}} V}} \tag{7}$$

and in the process of resonating it absorbs energy at this frequency. All the energy absorbed by the resonator during one part of the acoustic cycle is returned to the pipe later in the cycle. The phase relationship is such that all the absorber energy is returned back towards the source -- it does not get sent on down the duct. Since no energy is removed from the system, just returned, then the real part of the branch impedance $R_b = 0$. The imaginary part of the impedance may be expressed in terms of the compliance and inertia of the resonator, $X_b = p\,(w\,L_{\text{eff}}/S_b - c^2/wV)$, so that the sound power transmission coefficient may be written

$$T_\Pi = \left[1 + \left(\frac{c^2}{4S^2(\omega L_{\text{eff}}/S_b - c^2/\omega V)^2}\right)\right]^{-1} \tag{8}$$

The transmitted power is zero when $w = w_0$

in Eq. (7), which is the resonance frequency of the resonator, whence all energy is reflected back towards the source. This transmitted power is shown in Figure 4 for a Helmholtz resonator tuned to 172 Hz. These filters block any sound within a band around the resonance frequency, and pass all other frequencies. Thus they are called "band-stop" filters. The bandwidth of the filter may be increased by adding some porous absorbing material (steel wool) in the cavity of the resonator.

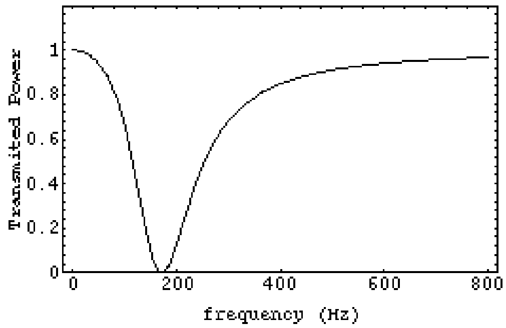
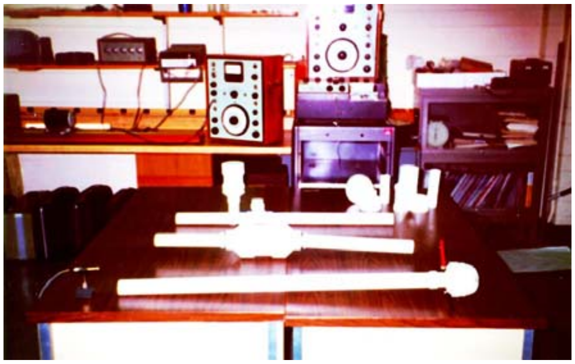


Figure 4: Sound power transmission for a band-stop filter tuned to 172 Hz.

IV. Equipment Needed

- PVC pipe:
  - a 1 m length (2-inch diameter) to serve as the unfiltered duct
  - several varying lengths of the 2-inch diameter pipe (instructor can pre-cut these lengths, or have students cut their own pieces)
  - several varying lengths (less than 40 cm) of 3-inch or 4-inch diameter pipe for building expansion chambers and Helmholtz resonators
  - T-joints, couplings between 2-inch, 3-inch, and 4-inch diameter pipe, end caps for all diameter sizes
- loudspeaker (good low-frequency response preferable) with connection to pipe (a horn loudspeaker driver fits rather nicely into a 2-inch diameter pipe)
- Frequency analyzer with averaging, multiplot display, and noise source, and floppy disk drive (Hewlet Packard 35670A)
- Audio amplifier to amplify noise input to speaker.
- Two small diameter (1/4") microphones and necessary preamplifiers



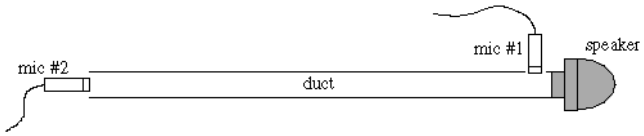
V. Procedure and Analysis

A. Setting up the analyzer and taking measurements

You will need a frequency analyzer to observe the frequency response of the duct as measured by the microphone. Set up the analyzer to display power spectrum of the microphone input over a frequency range of 0-800 Hz. If your analyzer has memory for saving data and the ability to display more than one plot or measurement at a time, you can readily observe the effects of each filter and compare the filtered sound measurement with the unfiltered duct. Save the unfiltered measurement to a data register and set up the analyzer to display the saved measurement and each new filtered measurement simultaneously (two separate displays or two measurements on the same plot). If your analyzer has a floppy disk drive you can save data to disk to plot later using a PC. You will also need a noise source to drive the loudspeaker. Using a noise source from the analyzer has the advantage that the frequency range of the noise source can be set to match the frequency range of analysis. For wave propagation down a duct this will ensure that only plane waves will be generated at the driver end of the duct. The noise source should be white noise (equal energy at all frequencies) or pink noise (equal power in all frequency bands).

B. Measurement of the duct resonances

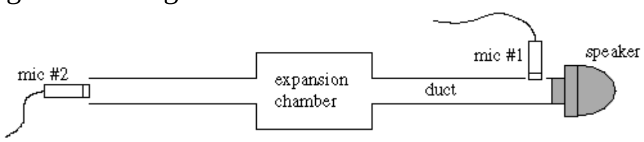
- Locate the long duct (approx 1 m in length). Be sure to record all pertinent dimensions of the duct.
- {Connect the 1 m duct to the loudspeaker and position the microphone about 1 cm from the center of the open end as shown below.



- Turn the source on and start a measurement. When the measurement is done, save the data to disk for plotting later. In addition, if you save this measurement to a data register you can display this saved data simultaneously with future measurements of the duct with various filters to see how the filters behave.
- Plot this data (or save to floppy disk for later plotting).
- Locate and record the values of any resonance frequencies of the duct. How do these frequencies compare to what you might calculate from Eq. (3)?

C. Low-pass filters

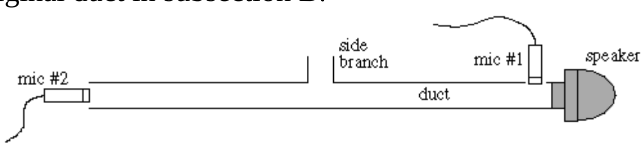
- Set up the duct with expansion chamber as shown below, but keep the total length of the duct/filter combination equal to the length of the original duct in subsection B.



- Be sure to record all dimensions for this system.
- Turn the source on and start a measurement. If you have set up the analyzer correctly, you should see this measurement superposed on top of the duct measurement.
- Did the measured response of the duct change? If so how?
- Plot this data (or save to floppy disk for later plotting)..
- Try a different size expansion chamber, but keep the total length of the duct/filter system the same.

D. High-pass filters

- Set up the duct with side branch as shown below, but keep the total length of the duct/filter combination equal to the length of the original duct in subsection B.

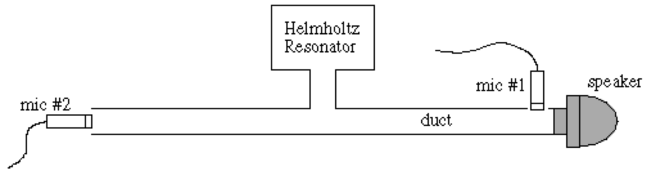


- Be sure to record all dimensions for this system.
- Turn the source on and start a measurement. If you have set up the analyzer correctly, you should see this measurement superposed on top of the original duct measurement.
- Did the measured response of the duct change? If so how?
- Plot this data (or save to floppy disk for later plotting)..

- Try changing the length of the side branch and describe what you hear. What length seems to be the best at cutting out low frequencies?

E. Band-stop filters

- Set up the duct with Helmholtz resonator as shown below, but keep the total length of the duct/filter combination equal to the length of the original duct in subsection B.



- Be sure to record all dimensions for this system.
- Turn the source on and start a measurement. If you have set up the analyzer correctly, you should see this measurement superposed on top of the original duct measurement.
- Plot this data (or save to floppy disk for later plotting).
- Did the measured response of the duct change? If so how? How does the measured value of the band-stop frequency match a calculated value of the resonance frequency of the Helmholtz resonator?
- Try to find resonators that will knock out each of the first two resonances of the original duct.

F. Ideas for further analysis

- Compare plots of the response of the duct alone to the duct with the various filters. Do the filters behave as the theory predicts? How so? or How not so?
- Comment on how such filters might be used in automobiles, air-conditioning systems, etc.

VI. References

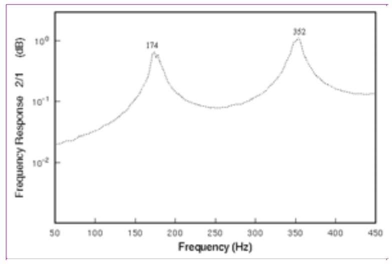
1. Kinsler, Frey, Coppens, and Sanders, *Fundamentals of Acoustics*, Third Edition, (John Wiley & Sons, 1982) Chapters 9&10.
2. Reynolds, *Engineering Principles of Acoustics: Noise and Vibration Control*, (Allyn & Bacon, 1981), Chapter 9.
3. Dowling and Ffowcs Williams, *Sound and Sources of Sound*, (Ellis Horwood, 1983), Chapter 6.

VII. Sample Student Data

This laboratory exercise has been used very successfully in a rather popular senior level course, "PHYS-580 / ME-530, Acoustics, Noise, and Vibration," which serves as an elective for Mechanical Engineering, Electrical Engineering, and Applied Physics majors at GMI Engineering & Management Institute. Students really enjoyed putting the duct systems together, and most experimented with "monster" filter networks to see what effect multiple filters have. The data shown below and on the following pages was obtained for a 1 m length of PVC pipe, 2-inch diameter with appropriate T-joints, 3-inch and 4-inch pieces, and necessary connectors. The duct was driven with a horn loudspeaker which fit nicely into one end of the PVC pipe. The general effect of high-pass, low-pass, and band-stop filters is rather clearly demonstrated for this system.

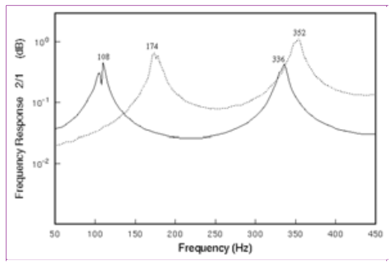
Measured response for duct, length $L=1.01$ m, radius $a=1.95$ cm

Resonance frequencies at approximately f_1=170 Hz and f_2=340 Hz

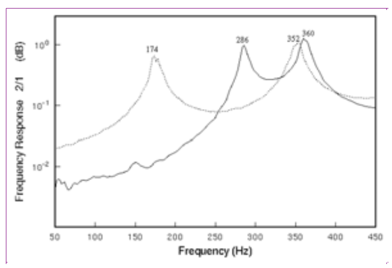


On the following plots, the dotted curve is the unfiltered duct response, shown above.

Duct with expansion chamber --- Low-pass filter

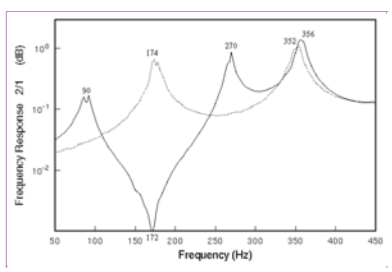


Duct with side branch --- High-pass filter



Duct with Helmholtz resonator --- Band-stop filter

Duct with Helmholtz resonator tuned to f_1=172 Hz



Thiele - Small Analysis of Loudspeaker Enclosures

This section is based on the classic work of Neville Thiele, Richard Small, and the earlier work of Bart Locanthi. It is a bit technical, and unavoidably requires some familiarity with electrical circuit theory. The first section presents a derivation of the equivalent circuit of a speaker in sealed, and vented (bass-reflex), enclosures. Then the relationship between the equivalent circuit response and sound level is presented. The effect enclosure volume has on efficiency, frequently misunderstood, is discussed. Then infinite baffle, acoustic suspension, and bass-reflex enclosure designs are compared. Additional issues regarding bass-reflex designs are briefly reviewed. Sensitivity of results to misalignment is discussed, and finally the interaction between enclosures, crossovers, and cables is investigated, as well as the effect of the amplifier damping factor.

Thiele-Small analysis overview

The Thiele-Small approach is to first analyze the electro-mechanical behavior of a speaker voice coil, magnet, and cone, interacting with the cone suspension and the air in and outside the enclosure. The resulting equation is mathematically identical to the equation describing a purely electrical "equivalent circuit" consisting solely of resistors, capacitors and inductors. The sound produced by the loudspeaker can then be obtained via a relatively simple circuit analysis. The highly evolved theory of filter synthesis can be used to adjust circuit parameters to obtain a desired frequency response. The parameters can then be translated back into physical quantities, such as enclosure size, to build the loudspeaker. This procedure provides a scientific framework (veneer?) for the art of loudspeaker design. A very useful result is that after the speaker has been assembled, the electrical impedance at the speaker terminals can be measured and compared to the theory. If it differs, the design can be tuned based on this measurement, which is both simple and accurate. The location and height of the impedance spike at resonance are sensitive to any errors in the design, as will be shown. (However, as noted below, for a bass-reflex design some shift in the location of the peaks can be caused by mutual coupling, rather than by a design error per se). This analysis is particularly valuable for designing bass-reflex enclosures where a ducted port (also called a vent) allows air inside the enclosure to radiate in conjunction with the speaker cone.

The driver parameters, which determine several of the equivalent circuit elements, are known as "Thiele parameters," and are fairly standardized; a table given here[13.5kb] provides precise definitions and units, and also points out some potential pitfalls in using published values. A method for measuring the parameters is given on the Subwoofer DIY page.

The notation in this section generally follows Small's June 1973 paper (he alters his notation between analyzing drivers, sealed enclosures, and vented enclosures, but all three situations are covered by the circuit derived here).

Back to the top

The equivalent circuit

Small's papers and Dickason's book skip the derivation of the equivalent circuit. You don't need this to build a loudspeaker, but I like to feel as though I understand what is going on physically, so I'm going to go through it in detail. Besides, I enjoy starting from bedrock physics. There are three basic equations:

1. the relationship between the voice coil motion and voltage induced by the driver magnet

2. the equation for pressure inside the enclosure
3. the relationship between forces and motion of the speaker cone assembly.

The cone is assumed to be rigidly connected to the voice coil. The analysis is performed for a sinusoidal frequency ω radians per second, and as usual can be generalized via a Fourier transform. All equations are in MKS units. The goal of the equivalent circuit analysis is to solve for the velocity of the cone, and velocity of the air in the vent. The far-field sound pressure produced by a known velocity is then obtained from equation P6 in the physics section on piston radiation. This equation is an exact solution for a flat piston mounted in an infinite baffle, and is generally a good approximation for real speakers.

Voltage induced by voice coil motion in the magnetic field

The voice coil of a real driver is ring-shaped, and rests within a concentric magnet structure (see illustration [8kb]). However the equations can be derived from a simplified rectangular structure shown here [3.5kb]. The voice coil is represented by the yellow wire loop in the simplified figure. The driver permanent magnet produces a field B, shown in red, pointing in the positive y direction. A coil movement in the +x direction is defined to be moving away from the enclosure interior. When the wire loop in the figure moves sinusoidally with a peak displacement x_C, applying Maxwell's equations (Stratton's equation 1, page 2) shows that the motion induces a voltage across the loop terminals given by

$$\text{(T1)} \qquad v_x = -j\omega x_c B \ell \qquad [kg - m^2 - s^{-2} - q^{-1}]$$

The length l is the total length of wire within the field, in meters. For a real driver using the common "underhung" design, the magnetic field covers the entire voice coil, and l is the total length of wire in the voice coil. For an "overhung" design, as shown in the driver illustration, l is the length of wire within the magnetic field. When a current I_C passes through the wire, it produces an additional voltage that must be added to v_X to obtain the total voice coil voltage v_C. This term will be added later.

Pressure inside the enclosure

The cone motion produces a change in the pressure inside the enclosure. A vent in a bass-reflex design also affects the pressure. The enclosure walls vibrate as well, but this contribution to the pressure is assumed to be negligible. A side view cutaway shows the geometry [2.6kb]. A speaker cone with effective area S_D moves sinusoidally with a peak excursion x_C. There is a vent of cross-sectional area S_V and length L_V. It is assumed that the air within the vent moves sinusoidally as a rigid mass with peak excursion x_V(discussion regarding this assumption later). Finally, it is assumed that the wavelength is large compared to the dimensions of the enclosure, so the pressure can be approximated as uniform throughout the interior. Then using equation 25, and equation 30 in the Physics section, applying the divergence theorem, and temporarily neglecting losses, it is found that the change from atmospheric pressure is

$$\text{(T2)} \qquad p = -\frac{\rho_o c^2}{V_B}\left[x_c S_D + x_v S_V\right] \qquad [kg - m^{-1} - s^{-2}]$$

where the air density ρ_O and velocity of sound c are numerically given by

$$\rho_o = 1.205 \qquad [kg - m^{-3}]$$
$$c = 344.4 \qquad [m - s^{-1}]$$

For an empty enclosure V_B is equal to the physical volume; filling the enclosure with acoustic material can increase the effective size. The theoretical limit of the increase is 40%, and Small suggests that the practical limit is 25%. Small shows that for a bass-reflex design frictional losses due to the air flowing through material in the enclosure, through the vent, and leaks, can all be represented by a vent loss term $-j\omega x_v R_V/S_V$. In other words, R_V represents all enclosure losses. The value of R_V is determined by a measurement of the enclosure Q as discussed below. This term must be appended to equation T2 for the bass-reflex case. For a sealed enclosure, these losses are lumped in with the suspension loss, and the vent loss term is omitted. The air mass in the vent is accelerated by the force produced by the interior pressure, resulting in a value of x_V given by

$$\text{(T3)} \qquad x_v = -\frac{p}{\omega^2 \rho_o L_V} \qquad [\mathrm{m}]$$

Including the vent loss term in T2, and substituting T3 yields the final equation for pressure inside the enclosure in terms of the cone excursion

$$\text{(T4)} \qquad p = -x_c S_D \frac{\rho_o c^2}{V_B - \dfrac{c^2 S_V}{\omega^2 L_V} - j\dfrac{V_B R_V}{\omega \rho_o L_V S_V}} \qquad [\mathrm{kg-m^{-1}-s^{-2}}]$$

For a sealed enclosure, the two terms involving S_V are omitted.

Forces and motion of the speaker cone assembly

The pressure in the enclosure produces a force on the cone given by S_D times equation T4. The force driving the speaker is produced by the current I_c flowing through the voice coil. For the simplified figure [3.5kb], the current traverses the magnetic field in the positive z direction, and produces a negative x direction force (Stratton notes that this force equation essentially defines "magnetic field")

$$\text{(T5)} \qquad F_{magnetic} = -I_c B \ell \qquad [\mathrm{kg-m-s^{-2}}]$$

The next forces are due to the cone suspension via the surround and spider. There is a spring force, specified by a compliance C_{MS}, and a frictional resistive force R_{MS}. As noted above, for a sealed enclosure R_{MS} is modified to include absorptive enclosure losses as well. Then the equation is

$$\text{(T6)} \qquad F_{suspension} = -j\omega x_c R_{MS} - \frac{x_c}{C_{MS}} \qquad [\mathrm{kg-m-s^{-2}}]$$

The resultant of the above forces acts to accelerate the cone assembly and air in front of the cone, defined as the moving mass M_{MS}. The final force equation is

$$\text{(T7)}\, I_c B \ell = x_c \left(\omega^2 M_{MS} - j\,\omega R_{MS} - \frac{1}{C_{MS}} - \frac{\rho_o c^2 S_D^2}{V_B - \dfrac{c^2 S_V}{\omega^2 L_V} - j\dfrac{V_B R_V}{\omega \rho_o L_V S_V}} \right)$$

Equivalent circuit components

Now the procedure is to convert the physical parameters in equation T7 into equivalent inductors, resistors, and capacitors. Equation T1 is used to substitute for x_c. Then the following definitions are made

$$L_{CES} = (B\ell)^2 C_{MS} \qquad L_{CEB} = \frac{(B\ell)^2 V_B}{\rho_o c^2 S_D^2} \qquad [kg - m^2 - q^{-2}]$$

$$\text{(T8)} \qquad R_{ES} = \frac{(B\ell)^2}{R_{MS}} \qquad R_{EL} = \frac{(B\ell)^2 V_B R_V}{\rho_o^2 c^2 L_V S_D^2 S_V} \qquad [kg - m^2 - s^{-1} - q^{-2}]$$

$$C_{MES} = \frac{M_{MS}}{(B\ell)^2} \qquad C_{MEP} = \frac{\rho_o L_V S_D^2}{(B\ell)^2 S_V} \qquad [kg^{-1} - m^{-2} - s^2 - q^2]$$

The components L_{CES}, R_{ES}, and C_{MES} are determined by the "Thiele parameters", and are therefore selected by selecting an appropriate driver. The driver also determines S_D and Bl. In practice R_{EL} is determined by the enclosure Q, denoted Q_L. Initially Q_L is typically assumed to have a value around 7, and is subsequently measured for the completed enclosure, and the design fine-tuned if required. The values of L_{CEB} and C_{MEP} are enclosure design parameters that may be selected to provide a desired response, and then the above equations are used to determine the corresponding values of V_B, S_V, and L_V. In theory there are an infinite number of vent solutions, but there are rules of thumb relating the vent area and length, given for example by Dickason. For a sealed enclosure R_{EL} and C_{MEP} are omitted from the equivalent circuit, and L_{CEB} is the sole free design parameter, which then defines V_B. Substituting these definitions into equation T7 yields the equation of the main part of the equivalent circuit of the driver and enclosure

$$\text{(T9)} \quad v_x = I_c \frac{1}{\dfrac{1}{j\omega L_{CES}} + \dfrac{1}{R_{ES}} + j\omega C_{MES} + \dfrac{1}{j\omega L_{CEB} + R_{EL} + \dfrac{1}{j\omega C_{MEP}}}}$$

Additional electrical components

as noted above I_c causes an additional voltage due to electrical resistance R_E, and inductance L_E of the voice coil. The full circuit also includes the amplifier, with its output impedance R_A, speaker interconnect cable with resistance R_C, and the crossover with resistance R_{XO}. Therefore the full equivalent circuit of the system is as shown here [6.5kb]. For a bi-amped design minor modifications are made to the circuit, and for higher-order crossovers additional components are added. The components of equation T9 do not physically exist, but I_c is the real current through the voice coil, and v_c is the real voltage across the driver terminals.

Back to the top

Sound output of the system

As stated in the beginning, the goal is to obtain the velocities of the cone and vent air. The amplifier is assumed to output a voltage with equal amplitude and phase at all frequencies. The equivalent circuit response is then analyzed, and the voltages v_X and v_V, as defined in the equivalent circuit schematic, are obtained. The physical peak cone velocity is then given by

$$(T10) \qquad u_x = j\,\omega\,x_c = -\frac{v_x}{B\ell} \qquad [m - s^{-1}]$$

The peak velocity of the air in the vent is

$$(T11) \qquad u_v = j\,\omega\,x_v = \frac{v_v}{B\ell}\frac{S_D}{S_V} \qquad [m - s^{-1}]$$

At this point, all references I have seen use the real part of the radiation impedance of a flat piston to compute the total output sound power. An exact solution of the radiation impedance of a piston, and a numerical solution for a cone, are given in the physics section. For a peak velocity that is constant with frequency, the real part of the impedance causes a power increase of 6 dB per octave in the "piston range," roughly where the speaker diameter is less than a wavelength. The power then levels out for higher frequencies. However the sound level on axis continues to rise at the 6 dB per octave rate because the sound becomes focused in a beam that gets narrower as frequency increases. In fact, for an ideal flat piston moving with a constant peak velocity, the sound level on axis grows 6 dB per octave for all frequencies, "DC to light" as engineers say. This is shown by equation P6 in the Physics section, the exact solution for sound produced by a piston in an infinite baffle, which is valid for any frequency. (It is surprising to me that this on-axis behavior is not mentioned in the references I have seen, or if it is I missed it). Although total output power is important, in my opinion on-axis sound level is more important. At a distance on 1 meter from the speaker, the sound pressure on axis for the driver cone and vent are

$$(T12) \qquad \begin{aligned} p_c &= -jv_x \frac{\omega\,\rho_o S_D}{2\pi B\ell} \qquad [kg - m^{-1} - s^{-2}] \\ p_v &= jv_v \frac{\omega\,\rho_o S_D}{2\pi B\ell} \qquad [kg - m^{-1} - s^{-2}] \end{aligned}$$

Neglecting mutual coupling, the two pressures are added coherently to obtain the total pressure, and the peak SPL level computed using

$$(T13) \qquad SPL = 20\log_{10}(p/0.00002) \qquad [dB]$$

Subtract 3 dB to get the RMS value.

In the initial version of the crossover design section I stated, without really thinking it through, that a speaker cone would produce a square wave by jumping forward, sitting still, and then jumping back. Equation T12 shows that the sound pressure is actually proportional to the second derivative of the position, the cone acceleration. To produce a square wave of sound pressure, the cone velocity must be a triangle wave. Since the derivative of a sinusoid (meaning a sine or cosine) is another sinusoid, a sinusoidal cone velocity produces a sinusoidal pressure, but this is an exceptional case.

Consider a simple case of a sealed enclosure, no crossover, and neglect the voice coil inductance. For an input voltage that is constant in amplitude and phase, the cone velocity peaks at the resonant frequency, and drops by 6 dB per octave on either side of resonance. There is a -90-degree and +90-degree phase shift well above, and below, resonance respectively. If the cone velocity were constant with respect to frequency, the sound pressure would increase 6 dB per octave. For the cone velocity as described, the sound pressure is constant above resonance, and drops 12 dB per octave below resonance. The sound pressure phase varies linearly with respect to frequency and distance from the speaker, which simply represents a time delay. Neglecting this, there is an additional -90-degree phase shift between cone velocity and pressure. Therefore the sound phase is shifted 180-degrees with respect to the input voltage. But assigning a label of "positive" and "negative" is arbitrary in the analysis, and reversing the labels yields the result that the sound pressure has zero phase shift with respect to the input voltage, above resonance. The bottom line is that the wave shape of the input voltage is replicated by the sound pressure, for frequencies within the operating range of a single driver. In general, the wave shape is changed dramatically by the crossover, and by multiple drivers.

Back to the top

Efficiency

Efficiency is defined as the ratio of radiated sound power to electrical input power. The sound power is equal to the square of the sum of the velocities (equations T10 and T11) times the real part of the radiation impedance, which is given by equation P11 in the physics section. The input electrical power can be defined as the real part of the product of the voice coil current and voltage, or it can be defined as the real part of the product of the amplifier output current and voltage, which will give a slightly lower value. Efficiency is a function of frequency. Within the primary operating region of the driver the efficiency is close to the "reference efficiency" defined by

$$\text{(T14)} \qquad \eta_o = \frac{\rho_o (B\ell)^2 S_D^2}{2\pi c R_E M_{MS}^2} \qquad [-]$$

T14 is identical with equation 31 in Small's June 1972 paper. The efficiency is a function of the driver parameters, but is not a function of the enclosure. In other words, for a given driver the efficiency will be the same regardless of the enclosure it is put in. Dickason also points this out. Yet there is a frequently reproduced graph from a different paper by Small that appears to show that efficiency decreases as enclosure volume decreases. What's going on here? Equation T14 can be re-written in a form where efficiency is proportional to the 3rd power of the enclosed driver resonance times the enclosure volume. So what is actually changing as the volume decreases is that the resonant frequency is increasing. It is true that for a fixed low frequency response a larger volume can provide higher efficiency. But you have to buy a different driver to achieve the efficiency increase.

Back to the top

Design comparisons

Sealed enclosures are normally classified as "Infinite Baffle" or "Acoustic Suspension." If the compliance of the speaker suspension divided by the compliance of the air in the enclosure (equal to V_{AS} divided by V_B in the table below) is greater than 3, Dickason classifies the design as acoustic suspension. The driver Q_{TS} should be less than 0.3 or so for acoustic suspension designs, and greater than this for infinite baffle designs. Bass reflex designs should have driver Q_{TS} values of between 0.2 and 0.5. There are rafts of possible enclosure design "alignments," corresponding to Butterworth, Bessel and other filters. In my opinion there is not a huge difference between the alignments, neglecting plainly bad designs. For comparison purposes I have selected three designs given by Dickason: an infinite baffle and an acoustic suspension second-order Butterworth, and a SBB4 bass-reflex. In selecting the parameters given below an attempt was made to produce a good result for each design. I found real drivers in the Solen catalog that are fairly close to the selected parameter values, except that the compliance used in the acoustic suspension design is about twice as large as any I could find. A rather compliant driver is desirable for this design. A value of Q_L=7 was used for the bass-reflex design. For clarity, the voice coil inductance is zeroed out in the following comparisons.

Parameters common to all three designs:

Nominal driver diameter: 12-inches Voice coil DC resistance: R_E = 6 Ohms

Amplifier, cable, and crossover resistance: 0.44 Ohms

Force factor Bl = 9.34 N-amp^{-1} Total system Q_{TC} = .707 Suspension loss R_{MS} = 3 kg - s^{-1}

Parameters that vary between designs

Design	Q_{ES}	Q_{MS}	Q_{TS}	M_{MS}	C_{MS}	f_S	f_B	V_{AS}	V_B
Infinite Baffle	.51	2.48	.45	.040	.71	30.0	47.1	.275	.187
Acoustic Susp.	.30	1.43	.26	.040	2.1	17.3	47.1	.824	.129
Bass-Reflex	.35	1.72	.31	.033	1.2	25.0	25.0	.475	.187

Infinite baffle compared to acoustic suspension

The infinite baffle and acoustic suspension designs were selected to produce identical responses, including an identical efficiency of 1.4%, and sensitivity of 94.2 dB SPL. The physical configuration is not terribly different in this case. Scaling the infinite baffle cabinet dimensions by a factor of .88 provides the correct volume for the acoustic suspension. Pressure inside the enclosure is inversely proportional the cube of the scale factor (see equation T4). Stiffness of a clamped panel is roughly inversely proportional to the square of the scale factor. So my guess is that wall vibration is somewhat more problematic for the acoustic suspension design. There is also less room for filling to absorb the back wave. As shown below, the acoustic suspension design is slightly more sensitive to misalignment. I don't see a major difference here, but unless size is very important, I would choose the infinite baffle.

Sealed compared to bass-reflex

The bass reflex design was selected to have the same cabinet volume as the infinite baffle. The efficiency is 2.0%, sensitivity is 95.8 dB SPL, so there is only a 1.6 dB advantage here compared to the sealed enclosures. The frequency response curve [32kb] also shows slightly better relative bass in the region 20-50 Hz - about 1.2 dB better at 20 Hz. The transient response curve [29kb] is a plot of the sound produced by a step-function voltage input; that is, an abrupt rise from 0 volts to 1 volt DC. The response of the sealed enclosure is tighter, but not grossly different (More on transient response below). Probably the biggest difference is in the peak cone excursion [35kb]. The RMS input voltage for this graph is 2.83, which is about 1.3 Watts. The peak cone excursion for the sealed enclosure is 1.7 millimeters. For the bass reflex the peak is about 1.1 millimeters until the frequency drops below 20 Hz, when it begins to shoot up. A rumble filter is really essential for bass-reflex speakers to prevent excessive cone excursions at very low frequencies. Otherwise the frequently quoted advantage of smaller cone excursion for vented enclosures is simply not true. The vent diameter in this design is a sizable 11cm (4.3 inches). The peak air movement in the vent at 20 Hz is ±19 millimeters (note that it is scaled down by a factor of 10 in the figure). This is over ±3/4 of an inch, for a relatively low power! Its not unusual for me to play music 10 dB higher than this, where the motion would be ±2.5 inches. At 20 Hz the air motion in the vent is about 12 times larger than the cone motion for the sealed enclosure.

Most of the difference, a factor of 5.5, is due simply to the ratio of cone area to vent area. There is a factor of 1.4 due to the higher sound output at 20 Hz for the vented design. Finally there is a factor of 1.5 due to the fact that the vent and cone are somewhat out of phase, so they have to work harder to produce the sound output. Consider the behavior of the sound in free space, 1 meter in front of the loudspeaker. At a very high SPL of 117 dB RMS the peak air motion is ± 0.4 millimeters at 20 Hz (this assumes far-field behavior and is obtained from equation 38 in the physics section). So the level of air movement in the vent is grossly higher than a "real" sound wave. The most serious problem is that the flow in a 4.3-inch diameter vent becomes turbulent for a peak excursion of more than 2.5 millimeters or so, at 20 Hz. So the vent flow is well into the turbulent region. The resistive force in the vent is proportional to velocity for laminar air flow, but proportional to the square of velocity for turbulent flow. Therefore turbulence in the vent creates distortion.

Back to the top

Additional effects for Bass-Reflex Enclosures

Mutual coupling between the driver and the vent alters the behavior somewhat, and is not included in Small's analysis. One effect is that the impedance resonant peaks are shifted from the values predicted by the theory. Locanthi reports that the upper peak typically moves down by 4-7 Hz and the lower peak moves up by 2-4 Hz. Jacobsen has studied the effect of mutual coupling on radiated sound pressure. Unfortunately the details are rather complex. The effect is typically largest near the vent resonance, where Jacobsen shows a difference of about 3 dB when mutual coupling is included. Even without turbulence the air in the vent doesn't move as a rigid mass. The effective length of the vent is actually larger than the physical length due to end effects. Dickason provides an equation for vent length that includes a correction. Dickason states that vent non-linearity is always a problem. Distortion due to turbulent airflow has already been discussed. Harwood indicates that for a 2000-ft^2 room the vent in the design example will produce significant distortion for sound levels above 95 dB SPL, which is not all that loud. Vents also can produce wind noise - hardly surprising considering the substantial air motion. Pipe resonances within the vent, and transmission of enclosure resonances can also be troublesome. Bottom line for me is that a bass-reflex design adds complexity and problems, without that much payoff in return.

Back to the top

Misalignment

Enclosure designs are actually quite fault-tolerant. For the sealed enclosure designs, reducing the box volume by a factor of 2 produces the changes shown in the table below. The nominal case is the correct design. The lower two rows are the result of cutting the volume in half. The first column shows the resonant frequency of the driver, and the second the relative response at 30 Hz. A bump is created in the relative response near resonance, shown in the third column. Ringing in the step response is indicated by the peak negative excursion given in the fourth column. Considering the gross change in enclosure volume, the changes in response are modest. The input impedance spike doesn't change in height, but moves to the resonant frequency given in the table, so the design error is easily observed via an impedance measurement.

Sealed enclosure, effect of halving the nominal volume

Case	Resonance	30Hz Resp.	Resp. peak	Step ringing
Nominal	47.1 Hz	-8.4 dB	.0013 dB	-0.20

Infinite baffle	59.4 Hz	-11.2 dB	0.67 dB	-0.27
Acous. susp.	64.3 Hz	-12.4 dB	1.07 dB	-0.29

For the vented enclosure design, reducing the box volume by a factor of 2, or reducing the port air-mass by a factor of 2, produces the changes shown in the table below. The nominal case is the correct design. The middle row is for half of the design volume. The lower row for half the port air-mass (note: due to end effects the length decreases by more than a factor of two when the air-mass is halved).

Vented enclosure, effect of halving the nominal volume and port length

Case	Resonance	30Hz Resp.	Resp. peak	Step ringing
Nominal	24.9 Hz	-6.5	0 dB	-0.28
Half volume	35.3 Hz	-12.5	1.11 dB	-0.35
Half port air-mass	35.3 Hz	-7.4	0.79 dB	-0.37

Again the changes are relatively modest. A graph of the frequency response [31kb] shows the difference of port and volume misalignment. A graph of the step response [33kb] shows that the misalignments do cause a significant increase in the ringing. Finally the impedance plot [35kb] shows that the misalignment is readily apparent.

Back to the top

Crossover, cable and damping factor effects

Crossover interaction with the enclosure

For the next cases the voice coil inductance of 1.7 mH is included. The infinite baffle sealed enclosure and vented enclosure are used as examples. A 4th order Butterworth filter with a 300 Hz crossover and no impedance compensation produces a significant bump in the frequency response [29kb]. A Zobel did not help. Bi-amping with the same crossover eliminates the problem. This interaction problem is similar to the one covered in detail in the crossover section. In my opinion the most interesting effect is on the transient response, as shown here [30kb]. The solid blue curve shows the result without a crossover. The 4th order crossover delays the initial positive response spike by about 2 milliseconds. The enclosure acts like a filter with an effective frequency of about 50 Hz, so it reacts much slower and is more significant for the transient response than the crossover, creating the negative response ringing. In the bi-amped configuration the crossover causes a slight increase in the ringing. Without bi-amping, it causes a lot more ringing. For the bass-reflex design the result is particularly bad. Minimum ringing produces a tight-sounding bass. A large amount of ringing produces bass variously described as tubby or muddy.

Effect of speaker cable resistance and amplifier damping factor

The results presented in this section include the effect of .04 Ohms amplifier output resistance, corresponding to an amplifier damping factor of 200, 10 feet of 16 AWG zip cord speaker cable with .04 Ohms per side, and .32 Ohms crossover resistance. (see the equivalent circuit schematic [6.5kb]). For the bi-amped configuration the crossover resistance is absent, and the circuit is more sensitive to cable and amplifier resistance. For this case, the effect of increasing the amplifier damping factor to infinity, and increasing the cable size to infinity is simply to produce an imperceptible 0.2-dB increase in SPL level. Did the infinite damping factor give the amp "greater control" over the transient response? No, the difference is imperceptible. So the effect of increasing the damping factor beyond 200 is insignificant. And the effect of cables is insignificant, unless the cables are a lot longer, and/or a lot smaller. I know many audiophiles will not believe this result. I would only say that there are many things that engineers do not understand well, but circuit analysis is not one of them.

Back to the top

Back to Design of the Sound System

* The Science of Beauty *

Audio Culture in the Nineties

by Herbert E. Reichert

For far too long, J. C. and I would work construction by day and make art by night. We would retire to our studios and paint. The next day at lunch we would share with each other how it went. I was always a little jealous because in addition to painting, J. C. would go and play his *very* vintage Telecaster in avant-garde downtown clubs like the Knitting Factory. I felt that just collecting records and playing them on my "hi-fi" did not honor the musical expression or demonstrate my gratitude for the glorious feelings that music provides. I longed to do more than listen to records and paint.

One night while eating dinner at a friend's house, I was distracted by the beauty of the singing coming from the living room. When I inquired about it, I was told "it is the radio". Later on, I discovered it was a Marantz Model 10 playing an all-Quad system. For days all I could think of was the beauty of that singing. I became determined to re-create that beauty in my home. I bought books on electronics and acoustics and began to try to understand why this singing had affected me so strongly.

The systems I heard demonstrated in the audio salons clearly did not have this effect. So I began to build my own. Soon I was spending all of my free time drawing schematics and soldering.

J. C. would watch me reading and drawing amp schematics at lunchtime and tell me I was going crazy. He would continue, "all anybody needs to enjoy great music is a great tape in their boom box ... forget all that audiophile stuff and spend your money on great tapes". Today, instead of doing construction, J. C. pays his bills designing high-end speakers and single-ended 845 amps.

After listening to Monk and John Zorn at my place, J. C. decided to build a triode amp and a pair of speakers to play his Sony Walkman through. Within weeks, he and I were spending the whole day at work talking "tube talk". UPS was bringing weekly deliveries of RCA 45's and 50's directly to the job site. We would be shaking, opening the boxes and holding the tubes up to the light, reveling in the beauty of the globe-shaped masterpieces of the 1930's.

The rest of the crew would roll their eyes. They would admit, when pressed, that these tubes were beautiful as "industrial art", but went on to say, "how can these dusty old globes outperform modern high-end gear at re-creating the sound of live music?" When we informed them that not only could these objects play music, but that they were capable of producing sensations of grace, beauty, and excitement on a level *beyond* today's hi-fi, they began to actively ridicule us. To them we had become "high-fever" boys, lost in retro-land.

One particular fellow worker, a carpenter and a fine artist, was especially active in his ridicule of us. Over and over, every day, he kept saying: "*It can't be worth it!*" It turned out that he was an active music lover with a wide-ranging record collection. Further conversations revealed that he was a BIG George Jones fan. So I invited him over after work to hear some country favorites.

When we got to my studio I asked, "Shall we start with *George*?" He gave me a big smile. As I set the tonearm down and turned up the volume I heard "He said he'd love her 'till he died ... " coming from the speakers. When I turned around this man had his hands covering his face and was shaking his head. "*It's worth it, it's worth it, it's really worth it* ... " he repeated through his hands. This man quit his job and invested his life savings to open a store in the SoHo district of Manhattan, selling pure triode amps and horn speakers. I know, this sounds like a man joining a cult, not a document of a musical catharsis.

The real subject of this essay is: What in the nature of reproduced music moves us profoundly and inspires a purgative cleansing of our hi-fi sensibility? How can a simple, inexpensive, 5-watt record playing system move people to change the course of their lives? I could tell you many more stories like the one above, or I could tell you that this was simply the result of using directly-heated triodes and horn speakers. I am afraid that would be an oversimplification and would take us away from the possibility of discovering what is really going on here.

I have been forced to ask myself: What aspects of this reproduced sound elicit the dramatic attitudinal and behavioral changes that I've observed? You must understand, that this type of reaction, this letting out of breath, this surrender to the music, this feeling of finally understanding what reproduced music can do to us, is not simply an equipment change, but a profound change of mindset.

Over the last decade, I have listened to most of the big commercial high-end systems (such as the Infinity IRS-V's, the Wilson Audio WAMM, the Apogee Diva, etc.) I have also been exposed to several Japanese-style triode/horn systems. These setups usually feature small triode amps driving Onken or Altec horn loudspeakers. The high-end and Japanese-style systems could not be more different in character. The differences are not in how much detail, how much spatial information, or how neutral the tonal balance is, but rather, in the essential character of the musical presentation.

When I asked my Japanese friend what he thought of the WAMM's he had just heard, he said, "*Giant robots are no fun*." Another Japanese friend said, "*Too serious*." I believe the Japanese prefer a system that "embraces" the listener like a friend or a lover. Kondo-San, designer of the Ongaku amplifier, refers to his "deep emotion" upon first hearing music played through silver foil capacitors. This is not the way Americans talk about their hi-fi experiences.

When the Japanese discuss the relative merits of an audio component they sound like they are talking about orchestra conductors. When Americans talk audio, it sounds like an optician's convention. Americans rarely mention their feelings when telling their friends about an audio component they have heard. What I am suggesting is this: the difference between these two audio cultures is not simply "taste" in hardware selection, but a fundamental difference in the sensibility that informs this "taste".

An audio system, like a painting or a novel, represents a series of decisions by the "author" to effect the final "work". The quality of the novel, painting, or audio system is the sum quality of all of these decisions. Each of these decisions has moral, ethical, and technical implications as regards the final product. The quality of all of humanity's creations resides in this moral, ethical, and technical decision-making. I daresay we can evaluate the character of an entire culture on the apparent quality of its decision-making. What we regard as a person's "taste" is really their character.

All of the members of a family, from one to many, and the ways they "support" themselves in society, all combine to make a little mini-culture. Whatever importance the making or reproduction of music in the home assumes is a reflection of the nature or quality of this mini-culture. In other words, it is the result of ethical family decision-making.

In Japan, ethics is taught as a subject in the elementary schools and a person's hobbies and interests are considered more important than what he or she does to earn a living. People are respected as much for their aspirations as their achievements. Fast-food chains compete for the most beautiful "bento", or lunch box. Ferrari's, Harley-Davidson's, and Western Electric audio systems are collected as masterpieces of industrial art. The Japanese have a firm grasp on form following function. In the U. S., I am afraid these lofty concepts are only rarely discussed.

I am not trying to suggest that one culture is superior to another, but that a distinctly materialist drift in recent American thinking has pointed product design and consumer taste in a direction where durability, originality, and beauty are more of a liability than an asset. I would also like to suggest that this has begun to change. By the end of the Eighties, high-end audio had become such a vacant form of conspicuous consumption that its lack of aesthetic, philosophical, or scientific underpinnings led the dedicated few to build their own.

When you build your own audio gear it is *imperative* that the choice as to what to build is informed by a philosophy and an aesthetic surrounding "what is important" to the builder. As a creator, the home builder becomes immediately involved in the aforesaid ethical decision-making. How it looks, how durable it is, and how much pleasure the audio creation gives are all in the hands of the creator. The most successful home builders find themselves in an ongoing engineering and aesthetic dialogue with others of their kind. These dialogues then connect together to create an audio sub-culture that soon begins to affect the consumer mainstream.

In Europe and Asia this home builder audio sub-culture is massive and highly developed. In America, during the Eighties, our own Do-It-Yourself culture almost died. A small handful of U. S. music lovers, repulsed by the sterile high-end sound of the Eighties, began experimenting and looking back historically to discover where audio went wrong. At first, this search was an engineering fact-finding mission, but it rapidly began to look like all the big pieces of an astonishing audio technology were in place by 1955.

In the late Fifties, just as audio was about to become a legitimate branch of engineering and science, the mentality of the consumer and the manufacturer shifted dramatically towards the small and the disposable. The effect of this shift has been to replace engineering creativity and an ethics of beauty with marketing and "the bottom line". The transistor pocket radios set the tone for consumer electronics for the next three decades. The "acoustic-suspension" loudspeaker and the possibility of light, cool, 100-watt (transistor) amps set the tone for the home hi-fi industry.

The present quality of products is in such decline that it behooves us to re-invent the radio, television, and home hi-fi. The coterie of serious home designers (not the parts-changers) had discovered by the mid-Eighties that there was more to music reproduction in the home than flat in-room response. These designers would ask themselves fundamental questions about what a home music reproducer *should* do. A few of them began to look at the Japanese and European hi-fi cultures and started to ask: Why are these systems so different and what are these music lovers listening for?

They soon discovered that not all audiophiles were satisfied with the "does it sound like live?" critique. Instead, these audiophiles would ask questions like:

Is the music thrilling?
Does the system convey moods or feelings?
Am I filled with awe at the artistry of the composer or conductor?
Does listening induce peacefulness and reverie?
Can music inspire great joy when played through the system?
What *range of emotions* is the system capable of conveying?

In other words, the communication abilities of the music reproducer are the first consideration. In America, we assume that if it were to sound "just like live", all of the emotional content would be conveyed *automatically*.

I submit this is a poor premise to base an entire industry on. There is no law that demonstrates or suggests a perfect reproduction of the original soundfield would convey *any* of the original artistic quality of the performance. Additionally, there is no parallel in other media. A photograph is no substitute for a painting. Motion picture film or video will not capture a theatrical production or sports event. Bronze castings of marble sculpture lose most of the original beauty. What makes us think that vibrating transducers will communicate artistic quality just because they are almost linear?

What is important to remember here is to remember that the record playing system is a media of its own like film, paint, or clay. With any artistic media, communication is achieved through the creative use of dynamic contrasts (drama) and profound architecture (form and structure). In this country, the creators have chosen to build audio that emphasizes the mental picture of the performance. Image, depth, transparency, and grain are of high importance to American audiophiles. Alternatively, the Euro/Asians have chosen qualities that emphasize the emotional content. These music lovers design for maximum dynamic contrasts, presence, vividness, and effortlessness.

This visceral approach to audio design, as opposed to the cerebral, allows for a direct experience of quality. Aspects of life such as God or beauty or love are experienced *directly*. No thoughts or measurements are required to prove these experiences ... we all know them from direct experience. Anyone looking at a Ferrari or a Rembrandt will experience its quality.

I am suggesting that Americans now begin to ask more from a home audio system. Rather than design systems that fool the ear (is it live or ... ?), let us design systems that move the heart. Let us experience the beauty and profoundness of great compositions directly. Any system will play a great *recording* well, but few audiophile systems will play the great recorded *performances* well. A truly wonderful hi-fi will force you to become involved in a great performance, *even if* it is poorly recorded. With these types of criteria, even very modest systems are capable of great beauty and excitement.

We can now see that music reproduction systems can be radically different in the character of their presentation, based solely on what the designer or audiophile decides is important. If we design for imaging, detail, and small size, that is what we will get. If we design for dynamics, presence, and beauty, we can have that too. However, our emotional response to musical program will be very different on these two types of systems. With this foundation it becomes possible to understand why audiophiles might choose horn loudspeakers and low-powered triode amps.

At equal acoustic outputs, as compared to conventional dynamic or electrostatic loudspeakers, horns offer a dramatic increase in dynamic capability, image size, and presence. Harmonic distortion drops to a quarter of the value found in audiophile direct radiator systems.

In contrast, most direct radiators severely compress dynamic contrasts and reduce image size to the proportions of a symphony on a table-top. These are both severe distortions for which there are no measurements. More importantly, these are distortions which reduce the fun and excitement of music.

When reproduced music lacks weight and body, when sudden transients fail to startle, and the lead singer is only two feet tall, what's left? Detail? Transparency? Tonal balance?

People often say that most horns "sound like horns" and are therefore "disqualified from audiophile consideration". To me, a 90% reduction in image size is a gross distortion, but owners of "mini-monitors" talk endlessly about imaging and transient response. But without weight and body, the transients fail to startle and lose most of their emotional power.

A system capable of reproducing an enormous soundstage, that showcases dynamic contrasts, and presents music with realistic presence, weight, and body will never fail to excite and arouse. These are the traits that the triode/horn systems use to communicate. These are the traits that stimulate our body and our unconscious mind. These are the qualities I believe must become an American engineering priority if American audio and home theater are to become vital and important "mini-cultures" of our domestic environment.

Few families can afford a regular diet of theater and concert tickets. When they can, most working parents are too tired to engage in dressing up, driving, parking, standing in line, etc. Consequentially, the popularity and importance of these stay-at-home "events" increases.

My experiences lately have shown the best triode/horn systems can easily exceed the movie theater sound of childhood memories. We all recall going to see James Bond, 2001, or the "spaghetti westerns" on the big screen. These are big wonderful memories. No one can ever forget the sound or excitement of these events. I think we all yearn for feelings like this. Live opera is the same, but who gets to go?

What I am suggesting is that the technology *already* exists to stimulate our hearts, our minds, and our bodies as nearly profoundly as the theater or the orchestra hall. What we must do is open our closed, ethnocentric audiophile minds and explore all of the technologies available, taking from the old and the new as it suits our purpose.

I have mentioned how the dynamic capabilities, low distortion, and giant soundstage of horns might contribute to the new "world-style" audio-visual systems, but I have said nothing about where triodes of the directly-heated variety can fit into this new type of audio. This is because by nature their low power is at odds with our goals. We are seeking effortlessness and dynamic impact; for these purposes, one can never have too much power!

With existing types of high-power devices, there is a catch. MOSFET's, bipolar transistors, tetrodes, and pentodes all have one thing in common: without global feedback, the distortion, bandwidth, and risetimes are severely limited. There are several tricks to reduce or eliminate global feedback but none of these topologies approach the simplicity of the directly-heated triode circuits.

Triodes can have open-loop bandwidths over 100 kHz, in addition to being linear and predictable in operating characteristics. The most obvious characteristics of directly-heated triode amplification are lack of dynamic compression, lack of confusion and congestion, and a feeling of purity that enhances the beauty of the individual musical lines. All of these traits appear to be related to the triode's ability to be fast and clear with *no negative feedback*. But they are still very low power. We get a little lucky here because horns are very efficient.

It is not surprising that three decades after the invention of high-power tetrodes and beam tubes and two decades after the invention of solid-state power devices, big American theaters were still using triodes to power their sound systems. The reliability, clarity, and impact of these triode/horn systems gave theaters like Radio City Music Hall and the Ziegfield no reason to upgrade.

My intent here has not been to revive ancient technology, or to discredit the ingeniousness of contemporary American audio designers, but to revive our sense of consumer and marketing ethics. I believe we have accepted too much advertising hyperbole and engineering dogma and it is beginning to cost us our musical souls.

It is time we stop and re-assess our domestic entertainment priorities. If we conclude that musical reproduction in the home is not improving and is in fact becoming sterile and unprovocative, then it is time to ask our audio critics and engineers to subscribe to a higher aesthetic and ethical ideal than the present one, which emphasizes photographic verisimilitude at the expense of emotional verity.

by Herbert E. Reichert

Return to article index
Return to main menu

Reconsider, Baby! - The promise of horns in the contemporary situation

by Joe Roberts

For being a totally out of the blue concept, single-ended triode amplifiers enjoyed a relatively easy ride to respectability. The stuff is cool, no doubt about it, and that doesn't hurt a bit. In fact there are lots of people who are big fans of triode amps who never heard them - they got whipped up into a frenzy just thinking about it. Many others who viewed these fleapower amps with suspicion were won over by the *experience* of good triode amps.

Triodes do all the things the mainstream audio institutions say good audio amps have to do extremely well. You want imaging, soundstaging, back wall, yak yak yak? Look no further. The triode is your best friend if 3-D is your illusion of choice. But the lesson in triode amps for the "high-end" is that there are a few things that the general run of audio amplifiers does not do, things that we don't even have words for yet. The way good triodes play music leaves many jaded audiophiles speechless when they hear it.

Then the project of building a system around one of these fabulous amps runs you right into the question "What do I use for speakers?" Good question, one of the eternal questions in audio regardless of whether you're running two watts or two thousand. It gets a bit trickier when you leave behind the established power and sensitivity norms of the industry and enter the domain of "experimentation." With five watts, you're on your own looking for speaks, homes - at least as far as the high end speaker world is concerned.

Believe me, eight watts from a 300B will play many available speakers to at least *medium* listening levels. People are more or less happily running ProAcs, Ensemble Reference 3As, Spica TC-60s, etc. After all, all of these people running around raving about SE amps are listening to them on *something*, no? As reported by CG in *Stereophile*, my two watt 45 amp played loud and proud on a pair of ProAc Studio 100s.

The *subjective*
sense of power that a triode amp can deliver far exceeds expectations. It is not unusual to hear reports that this or that >10W amp sounds louder, fuller, and weightier than this or that 100W transistor amp or p-p pentode amp. Whether a particular speaker will work on a particular amp is a question for empirical research. A high and flat impedance and >90 dB sensitivity helps.

From *my*
listening chair, I say power ain't nuthin but a number as far as musical and emotional impact are concerned. But my chair is in front of a huge pair of high-efficiency speakers: 15" Altec woofers, Edgar midrange horns, Gauss compression tweeter. The listener with 8 watts and an 88dB speaker will encounter limits. How constraining these limits are depends on your needs. If you really like to crank up your stereo now and again or large scale orchestral music is your passion, the speakers you already have probably won't really shake the rope on three watts.

Triode amps sound great turned up to realistic SPLs, which makes the whole situation even more tragic.

ENTER HORNS

Thanks to the collective search for good high efficiency loudspeakers to use with low-powered amplifiers there's a lot of curiosity afoot about horn loudspeakers, probably the most unfashionable topic known to the modern high-end. Even 8 track gets more (and better) press than horn speakers in the US specialist mags.

Once upon a time, in the post WWII era, the earliest balls out hi-fi systems were constructed from recycled theater gear. Through the 50s and into the 60s, swanky top-of-the-line home hi-fi speakers featured arrays of horn-loaded squawkers and tweeters. A lot of this plastic horn and phenolic diaphragm stuff is a bit rough on the modern audiophile ear, but it sure was cool to be a "horn man" back then. Ain't that way now.

The need for high sensitivity dwindled as transistor power multiplied. Designers focused on ribbons, electrostats, and miscellaneous cones in sealed boxes intended to be driven by banks of steaming transistors or big hog parallel 6550 amps. The pioneers of the "high-end" as we know it had a different listening program and horns didn't fit. Quite a switch from the McIntosh and JBL 1950's upper crust hi-fi mentality. The horn ceased to exist in the mind of the modern US audiophile, except as a bad joke.

This is where triode amps and horns differ: single-ended amps are something totally new in the Western audiophile cosmos, horns have been tried and rejected. Nothing does more damage to music than a bad horn system and everybody knows it.

You can get a lot of music out of a good triode amp and a *good* horn. But, of course, it's not that easy. One minor problem is that *good* horns are extremely uncommon. Most horns are *totally unlistenable* in a serious music listening context. In my opinion, the bad reputation horns have endured among latter-day audiophiles is largely deserved. Most horns are junk.

But it is obvious that the kind of sensitivity a horn can provide would sure come in handy when you've got three watts and some change to burn. Plus the kind of crazy romantic audio nuts adventurous enough try a single-ended amplifier just to see what it can do are just the type who would gamble on horns too. So, here we are giving triodes and horns, the cutting edge technology of the thirties, a try in the Pentium age.

THE RISK OF TRIVIALIZING THE ISSUE

Because most horns are awful beyond description, if the goal is to pick up a few tricks which will lead us toward more perfect reproduction systems, we must be very selective. Most horns will be a total waste of time. If it looks like a cheap piece of junk, it is. Any horn made out of thin cheap plastic or cast metal that rings like a bell is going to be a problem. After all, how many cheap thin plastic or cast aluminum musical instruments can you name?

Our forebears figured out that most of those bottom of the line Jensen and EV $6.98 horn tweets were junk back in the fifties. We don't need to go through that discovery procedure again. Because horns vary so wildly in quality and performance, there is a real risk in thinking a "horn is a horn" and leaving it at that. While an "average" triode amp still sounds pretty decent, an "average" horn will DESTROY MUSIC.

Only the top 10% of the horn population is worthy of consideration for serious music listening. A few lesser horns are okay to play with for fun and may do some really interesting things, but they will have at least one dire failing which will have them in the garage after a few weeks playing time. Leave the junk in the airport paging systems where it belongs.

In recent attempts at covering the "Horn and Triode Scene," whatever that is, one big reviewer hooked up some cheap and junky squawk boxes with popularly priced SE amps and, in essence, reported that "Wow, this is better than I thought it would be but it's not that great, *really*." The writer was obviously having a good time, which is fine by me, but I don't think that he exhausted the possibilities of the horn genre with that experiment.

Screwing around with some funky cheap speakers is great recreation - I do it every chance I get - but it will not provide a lasting contribution to the goal of achieving real magic in the listening room.

What is needed is a *profoundly deviant* mindset. We have to search for something *way better* than what we've got to get trivializing the issue. A serious dialogue on horns, centering on open-minded evaluation of the finest of the species and reevaluation of our present-day goals and achievements, might get us to new and exciting places.

Horns and Mainstream Aesthetics

One day I was discussing the marketplace realities of horns with Peter Qvortrup of Audio Note UK. He evaluated the situation thusly: "There is so much suspicion about horns in the marketplace that listeners will be against the product before they even hear it. If you go out on the market with a product that doesn't do what they want and *expect*, you'll get destroyed".

Come to think of it, there is a bit of a party line regarding what constitutes "good" in the audio mainstream, a standardized aesthetic program. So what if horns do some musically relevant things better than cone and planar "high-end" speakers? Even the very best horns I have heard do not do some things on the 1994 Official Audiophile Speaker Criteria List.

Peter's findings were that horns do not give you "hall sound" and he predicts that reviewers will freak out if there is no "hall sound" regardless of how right the speaker is otherwise. Minus this critical performance factor, there could be PR problems ahead for our old friend the horn speaker.

I personally think "hall sound" is a cool illusion. Hall sound, as I understand it, is a recreation of the original acoustic space in which a recorded performance took place. This is, of course, partly a matter of the quality of a stereo recording, but a reproduction system has to be tuned in the right way to furnish an illusion of the original recording space. It's become a given that good systems do all this "hall sound" stuff right and a lot of contemporary audiophiles like this phenomenon. Most, I am sure, never questioned the issue since it sounds so good on paper.

Granted that hall sound is a neat illusion, isn't it strange that when we are supposedly trying to get to the sound of live music, we evaluate what we hear in terms of "soundstaging" and "imaging"? These concepts are strictly audio geek notions, totally irrelevant to the experience of live music. The language some people use to talk about reproduced sound suggests they are more concerned with questions of architectural acoustics of the hall than the music on the stage.

While it is nice to sit in a great hall and hear some unamplified acoustic music, it is unnatural to focus on the hall sound while you are there. You came to hear the music, right? Nobody talks about music that way and, as far as I can tell, live music doesn't even do most of the 3-D stuff that audiophiles insist on from their systems.

Furthermore, the effect of imaging is contrived and fake even when done well. I loved my Spica TC-50s but there wasn't any way they could reproduce any music except solo mandolin with a realistic sense of scale. Would you accept what passes for good hall sound in your speakers as good sound in an actual hall? No way, Jose.

Far from being an *absolute*
part of listening to music, all this a soundstage jazz is an acquired taste and a rather obscure one at that. I like it quite a bit myself, but it took me a lot of magazine reading back in the Seventies and Eighties to even figure out what the writers were talking about.

Concepts like "dial in the soundstage with some Shun Mook ebony root pods from Mother Africa" are not intuitive. Ever notice how your non-audiophile friends never volunteer that your imaging is superb or remark, "Wow, I never heard the back wall on that recording before."

If you have to *learn how to hear* this stuff and *learn*
that it is important, this suggests to me that hi-fi is not directly realistic despite the claims of the orthodox ideology. What we consider "real" is a matter of agreement rather than anything "absolute". There are codes and styles in reproduced sound just as there are in "realistic" visual art.

They are HERE

Contrary to popular folklore, horns can project three dimensional images. They are still "better" images than you get live. Big weighty images that grab you with presence and impact, just like real music, rather than relying on unnatural levels of hot top end "detail". 3-D is no problem for good horns.

Horns have a very forward presentation. Back in the seventies, "too forward" was a common criticism of speakers. What people were looking for was that backward sound, I guess.

The illusion horns provide is a "they are here" sound rather than the old "you are there" illusion. That is, the sound is so dynamic and alive that it sounds like the music is going on IN YOUR ROOM. True, the "soundstage" illusion of reproducing the original hall sound is skipped over as a consequence. After a few years of listening to horns, I strongly prefer the "right here in the room" sound of horns to the "looking into the room from a hole in the wall" sound I used to get with mini-monitors.

My real complaint about most soundstage projecting speakers, and I haven't heard them all, is that they can't rock out. Sometimes I just gotta listen to some old Funkadelic and whatnot. Stage boundaries are irrelevant in this context.

Maybe there is some grand cosmic tradeoff between dynamics and soundstaging. I think a lot of the hall sound master speakers create that unitary soundspace illusion by flattening out the dynamics somewhat and confining the presentation to a small listening window.

How can a 6" cone and a dome tweeter produce a powerful and spatially grandiose recreation of an orchestra? Most of the instruments you are trying to reproduce are way bigger than these dinky boxes. How can you expect "real" sized piano presence or the visceral growl of a bowed cello in front of a 6" plastic cone? Forget it. Getting the dynamics right and a larger sense of scale creates a stronger illusion for me than fake soundstage information, even on well-miked acoustic music.

In Search Of The Perfect Jukebox

Horn lover and audio philosopher Dennis Fraker nailed it down in a recent phone chat: "What audiophiles want is a really good jukebox, except they don't know that's what they want because they never been exposed to it and people are telling them that they want something else. But when they hear it, they know that's what they really want. You know, the right jukebox in the right bar can be magic..."

Whew, that phone call went off on a wild tangent, but he had a point there. Back when I was a snobby, sweater-wearing "high-end" salesman I often used the word *jukebox* as a term of derision, as in "the IRS ain't nothin' but a rich man's *jukebox*." Years in the hobby later, I realize that the perfect jukebox is perhaps the loftiest goal a music listener can aspire to attain. And perhaps the most challenging and difficult goal as well. The perfect jukebox, think about it.

If you're looking for buying advice, I can't give you a turnkey horn package purchase recommendation yet. At this stage of my explorations, I have enjoyed glorious results with my enormous 3-way Edgarhorn based system and I found a few "parts" that demonstrate real promise for music listening. The Stage Accompany Compact Driver is one such intriguing "part" (*see sidebar*).

After unpacking a loaner pair of Compact Drivers, I popped them on top of my Altec 416/Onken bass cabs, connected a 10 mF Hovland MusiCap speaker cap in series to roll off the driver around 1 K, stuck a coil in the woofer line, and turned on some music. Nice! LOUD! CLEAN! That was easy. 10 minutes down and we were rocking.

You know how well done live amplified music has that immediate sound that you can't get in your home even though both both are products of amps and speakers? Well, the Stage Accompany sounds like a great sound reinforcement speaker. It's the kind of speaker you hope the jazz club has when you go down to hear Betty Carter and it can bring Betty into your home for your listening pleasure.

Reminds me of a discussion I read on the Internet one time. One sage hobbyist wrote something to the effect that "of course, blues music sounds great on paper cone speakers and tube amps - it was produced on paper cones and tube amps!"

Conversely, if your thing is music that went through a microphone or pickup at the live venue, maybe something like the Compact Driver is what you need. It is musically satisfying to listen to a speaker that can slice through ether the way a Hammond B-3 played through a Leslie cabinet does. Leslies don't "image" and "localize", they energize the whole room.

The ultra clean sound of the Compact driver really helped out acoustic music too. Listening to Ralph Stanley transported me right back to old Virginia. The banjo notes sprayed out like ice and the fiddle breathed and moaned. "Natural" is a good way to describe the sound of this ribbon speaker and I'm not using this word in the metaphorical audiophile sense here.

My experiments with the Compact driver really showed me some new possibilities in sound. What more could you ask from a component than a glimpse of the beyond? It may be years before you can go out and buy a "high-end" approved horn system. Might never happen. In the meantime, listen and grow.

1)
One obvious solution to the power challenge is to BI-AMP, preferably with crossovers before the amps. Use that SE triode amp on the mids and/or highs and use your Jadis, ARC, VAC, Aragon, or whatever on the low end. I resisted the concept of multiamp systems for years because I thought it was too complicated to work. I was dead wrong. Using different kinds of amps in roles where they'll perform best makes sense to me and it gets around the low power issue entirely.

Dionomite!

At $25,000, Be Yamamura's full-range, single driver, multi-horn Dionisio is unlikely to show up at your local high end saloon, or in my system, for that matter. But after listening to a cone-driven Edgar mid horn for a few years, I am a champion of this concept. At a range of ten feet, metal midrange drivers can blast a hole in the wall behind your listening position. A horn loaded cone can really sing close in.

The Dionisio uses a modified Lowther PM-4 to drive this 2.3 m tall cork covered fiberglass sculpture between 27 Hz to 16kHz with better than 100dB efficiency. Said to play on two watts.

Yamamura has various smaller and larger versions of the Dionisio in the works, some priced down in the economy car range. Crazy price aside, paper cones, cork, and no crossovers sounds like a reasonable recipe to me.

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Novato, CA 94948
415-898-8067

Once Heard, Never Forgotten -

so goes the motto for the Turbosound line of top-shelf pro sound speakers.

Next time you want to hear Dark Side of the Moon, wheel in a pair of Turbosound monitors and put the LS3/5As in the closet. If they're good enough for Pink Floyd concert performances, maybe they're good enough for our home systems.

The TCS-612 system pictured above is set up for bi-amping with an electronic crossover. LF sensitivity is 98 dB @ 1m/1W and HF sensitivity is 103 dB @ 1m/1W. Response +/- 4dB 60-20000 Hz.Nice comfortable 16 ohm load. Size is manageable: approx. 23H X 14.5 W X 14.75D. Midrange uses a horn loaded 6.5" paper cone with concentrically mounted titanium compression driver twweter. The 12" bass is reflex loaded. Turbosound also makes some mighty impressive high power systems with modular components in the 106-108 dB sensitivity range.

I haven't talked with anybody who evaluated this stuff with triode amps in a home listening situation but it sure looks promising in theory. Can audio nirvana be found in the pages of Mix magazine?

The Compact Driver is a ribbon driver with a neodymium magnet. Unlike the plumbing fixture pro horns of yore, this unit w find this level of performance for $12.95 from a catalog featuring VCR belts and Pyle Drivers, think again. US distribution: Stage Accompany USA 4106 Fox Run Trail 6 Cincinnati, OH 45255 513-528-4035 513-528-4037 fax Nominal Impedance 12 ohms flat Frequency range 1-30 kHz Peak power 1000 W peak /60 W cont. Sensitivity 103 dB @1m/1W Max SPL (127.2 dB @1m/1W) Reprinted from SOUND PRACTICES #7. Editorial/subscription info: sp@tpoint.com

Blue Thunder

by R. Luigi Andreoli and Christian Rintelen

It all started with commercially available, 94 db/W/m-speakers and a custom built triode amplifier. The virtues of this combination were evident - but so were its flaws. Different component swaps showed that the speakers were the weak link in the chain - as always. We looked around for a speaker that improved the strenghts and reduced the flaws. But no commercially available speaker met all our requirements - they were either to big or to ugly or coloured to much or did not fit to the amp or In short: We decided to start from scracth and build a speaker exactly to our taste.

What do you want?

Building a speaker is always dealing with compromises. The no-compromise-speaker does not exist, despite all claims by manufacturers. Satisfaction has a lot to do with defining goals and being fully aware of compromises. There will still be a lot of surprises to handle during the process, but one better starts with fixed goals. Otherwise you will never arrive.
We set ourselves the following targets for our speaker:

. it should be a true full range-speaker with as little drivers as possible, preferably two way
. it should be of high efficiency (SPL over 100 dB/W/m)
. it should be easy to drive for triode amplifiers (loads, resonances)
. it's sound should be powerful and dynamic, yet natural, detailed and as uncolored as possible
. it should not only please our ears, but also our eyes
. it should work well in medium-sized (i.e. 30 m2) rooms
. it should not require a listening distance of more than 4 meters

As we could not use the room corners, we had to forget about corner horns like certain Klipsch or Lowther-designs.

Which way to go?

Our goals were high and the restrictions clear - we thus decided from the beginning on that we were prepared to invest in the best available components. The demand for high efficiency and pleasing esthetics narrowed the choice down to horn loaded bass systems or Onken-like cabinets. Going two-way all the way on the other hand required a driver that was could manage all frequencies from 500 Hz to 20'000 kHz. Futhermore, it should be horn loaded to assure homogenity with the horn loaded bass.

There are not many drivers of high efficiency that fulfill these requirements. Evaluating the different possibilities showed that one driver was perfectly up to it's task, the TAD 4001 compression driver. This 2" unit with a beryllium diaphragm has a flat frequency response from 500 Hz to well above 20 kHz an a nice 12   impedance at the crossover frequency range. Efficiency is around 108 dB/W/m, due to the almost 3 kilograms of Alnico-magnets. The compression driver weighs a hefty 13 kilograms.

Designing a horn that covers such a wide frequency range with even sound dispersion and without high frequency roll off is a tricky job. We thought about building our own horn, probably using a Iwata-curve or some other exotic design. Doing so would cost a lot of work and money without guaranteed satisfaction. TAD on the other hand offers a horn especially designed for the 4001 compression driver. We decided to go for that option, assuming that the guys from TAD surely know how to get the best out of their drivers. After all, their driver/horn combination is well reputed in recording studios all over the world.

The TAD's horn width is 61 centimeters - a measure of some significance, since we didn't want the speaker to look like randomly piled up boxes. The original Onken design is 80 centimeters wide, putting the TAD horn on top of it would have been a visual mismatch. In short: The cabinet of the bass speaker should also be 61 centimeters wide. This ruled out any Onken-style cabinets with 15" woofers, since the area of the reflex openings on that design is equivalent to that of the driver surface. Knowing that the Onken was actually designed to be used with transistor amps (even the models with the reduced vents are reported to have a slightly lumpy bass with tube amps compared to transistor-designs), we decided against it.

Comparing the different driver options for the woofer, we decided to use the Focal Audiom 15 VX 2 instead of the Altec 416, since the latter needs a 30% bigger enclosure volume for the same bottom end. Horn-loading down all the way was out of question, since the opening required for 25 Hz would be huge. So we settled for a vented cabinet with horn loading down to approximately 120 Hz.

Tractrix or exponential?

The tractrix curve is said to produce good sounding horns in the 150-500 Hz range. In addition to this, a tractrix-curve would allow a nice visual effect by using the cabinet walls in 45 degree-angle to blend perfectly with the miter joints we intended to use for the cabinets. (By the way: The last 20% of any tractrix horn are useless - the cabinet walls made up exactly for this percentage!)

The Audiom driver needs a volume of about 220 litres in a vented cabinet to produce bass down to 40 Hz. Add that to the volume of the horn plus the massive plywood - and you get a fairly big cabinet. With the front being relatively narrow, the finished speaker looks impressive but not intimidating or dominating. The finished speaker measures 61 cm w, 125 cm H and 70 cm D (including midrange horn). Weight is approximately 150 Kilograms. Finish is clear laquer for the enclosure and dove-blue nextel for the tractrix horn and the crossover box.

Crossover-frequency is 560 Hz with a 12 dB-design using two huge coils in the bass to minimze DC resistance. We chose a symmetrical configration since the output transformer of the amp is not grounded. This allows to drive the woofer on both leads equally. The crossover for the TAD can either be used in conventional configuration for single-amp-operation with a stepped attenuator providing 0,3 dB-steps or with the attenuator bypassed for bi-amping. The choice of excellent drivers allows simpler crossover designs since you don't need extensive filtering to fix any inherent ditches and flaws.

The Audiom woofers were burned in" (i.e. maltrated) for three days with different frequencies before mounting. Despite this, the bass reproduction continued to improve significantly for half a year!
We selected 30 millimeter birch-ply for the cabinet in order to reduce resonances. The tractrix-horn was made of 25 mm MDF - a wise choice regarding the many odd angles of the tractrix horn. The cavity between tractrix horn and cabinet was filled with sand.

The sound

Like everybody, we wanted a speaker capable of credibly reproducing all kinds of recorded music. To us, this credibility" includes (and among other criteria):

1. convincing micro- and macrodynamics
2. timbral correctness
3. no disturbing colorations in any frequency range
4. credible reproduction of soundstage depth and width
5. authority (the neccessary body and air real music has).

Just a brief description of what we ended up with - as unbiased as possible:
The speaker produces a big, emotionally involving, but nevertheless precise and thus credible reproduction of the recorded event. It does not turn a lousy recording into gold, but already mediocre recordings and pressings really shine. The bottom end has tremendous energy and authority. It is fast, colorful, tight and controlled without (audible) cabinet resoncances. It really moves air when kickdrums are kicked. The far left of a piano has the neccessary attack and speed to be considered almost real". Double bass is

reproduced sonorously with plenty of snap and color. The midrange is seamless and very homogeneous, despite the crossover in the critical 500 Hz-range. Harmonics of bass instruments really sing". Forget any prejudice about horn speakers not being suited for voices - the TAD sure is. The highs are neutral, extended and sweet. Hot or bright recordings can be tamed by the very fine steps of the attenuator. What is especially pleasing with this two-horn-speaker: You don't have to move 20 feet away to get homogeneity, i.e. seamless blending of the two horns - 10 feet are enough.

The conclusion

We are happy with the result. Of course, the speaker is big. But it integrates well into rooms with 30+ m2. The money spent on excellent component quality was well invested. Our recommendation to all DIYers: Don't build a big speaker with little money - build a small one with the best components you can afford. It certainly pays out!

Room for improvements

Remember: The perfect speaker does not exist! A fully horn loaded bass for instance would speed up the bass even more. A supertweeter above 15 kHz would probably add some extra air and reduce directionality of high frequencies. Birch-ply sounds very nice and is ideally suited for this kind of speaker - but a real pain to work with Be prepared for more!

The authors

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Pictures of the Blue Thunder speaker system:

Front View of Blue Thunder
Internal view of Blue Thunder
Side view of Blue Thunder
The crossover unit for Blue Thunder

Why Horns??

So why would anyone want to use horn speakers today, when we can have 100W amps at low costs? As I hinted in the previous paragraph, horns DO have other special qualities than just high efficiency. Okay, it's necessary to understand a few basic things about HOW HORNS WORK in order to say WHY they do some things better than other speakers. (You'd be amazed at how many people who simply refuse to believe that horns can be used for serious music listening because somebody told them so or because they think horns all sound like megaphones.)
A horn can be viewed as an acoustic impedance transformer. Turning mechanical motion (a vibrating diaphragm) into sound waves in air is in many ways a difficult thing to do. The most fundamental problem, which has a lot to do with the issue of efficiency, is that the difference in DENSITY between a paper or metal diaphragm and AIR is huge. There is a tremendous impedance mismatch. This fact explains that sound travels very far through denser media like metal, water or rock. In a speaker-air situation, the speaker diaphragm can be seen as a high impedance source (solid material) and the air being a low impedance medium (the air does not easily load down solid moving objects). There is a reason why humans can't fly by waving their arms!
What the horn does is to help the transducer couple its radiated energy into sound waves in air by means of an impedance transformation. What this means is that it creates a higher acoustic impedance for the transducer to work into, which means that more power is transfered. (Analogous to putting an antenna on a radio transmitter, which seems like an obvious thing to do!) Basically, a horn is a tube or conduit with increasing cross-section along its axis. The narrow end (where the driver sits) is called the horn throat, and the large end (which opens into the room) is called the horn mouth.
Sound pressure is defined as pressure change per unit area. In a horn, the wave front is restricted by the inner walls of the horn, and the area across the horn increases as the wave front approaches the horn mouth. So what happens here is that at the throat we have a small area and high pressure with small amplitudes, efficiently loading the diaphragm. As the wave fronts travel towards the horn mouth, the pressure drops, while the amplitude and the area increases.
A horn also has the property of directing the sound into a narrower beam, which increases the on-axis sensitivity (SPL/1W/1m). Increased directivity combined with high electric-acoustic conversion efficiency means that horn speakers are very easy to power, even with very small amplifiers.
What does all this really mean, then? In what ways does the horn 'help' the driver/transducer. And how does all this make horn drivers a bit different from direct radiators? I will try to sum this up in a few points:

-Improved energy conversion means that for a given SPL, a horn loaded diaphragm will have to move less than a direct radiating diaphragm of equal size. For any electromechanical transducer, the distortion generated by the driver itself will be proportional to diaphragm excursion magnitude. Thus, for any given SPL, the horn loaded speaker will have lower distortion than the same size direct radiator.

-A smaller diaphragm on a horn can be used to generate the same SPL as a larger direct radiating diaphragm for the same excursion amplitude. This means that you have a smaller mass to accelerate for the same acoustic output when you horn load a driver. This helps the transient response of the speaker regardless of what Fourier said. Subjectively, horns will be noted for their effortless, snappy handling of transients.

-The smaller diaphragm excursions allow the use of short, underhung voice coils (reduced mass again) taking full advantage of the flux in the pole piece gap. This increases the efficiency of the transducer, allowing the amplifier to work with more headroom and greater ease. Horn drivers need to have powerful magnets and tight magnetic coupling because of the high pressure they are asked to produce when sitting in a horn throat.

-Because the amp has more headroom, and because the driver handles signal peaks and high outputs more ideally, horns will be able to produce much higher SPLs than comparable direct radiators before distortion becomes objectionable. In short, there will be room for more dynamics, at lower distortion, with better transient response, with less stress on the amp.

Since this is an enthusiastic pro-horn text, I have not emphasized the problematic aspects of horns. One thing I haven't mentioned is that the lower the frequency one wants to reproduce through a horn, the larger the horn must be.

The size of a horn quickly multiplies when you go down a few octaves. Bass horns can be next to impossible to fit into a normal home. This really shouldn't be a problem to a true enthusiast, but even I have had to postpone any dream of a bass horn until I get a bigger place to live.

Some people say that horns have 'horn sound'. I'm not sure but I think what they mean is a sort of megaphone-like quality to the sound. A good horn should not have any of this. Unfortunately, many bad horn designs have led people to think that this is how horns 'are supposed to sound'. To the true horn fan, 'horn sound' will be a compliment that means, clean, dynamic, 'fast', physical, detailed and present sound.

Admittedly, there are a few things that horns don't always handle quite as well as your typical small direct radiator speakers. Particularly the much hyped concepts of 'imaging', 'neutrality' and 'transparency'. Horns will often lean towards a more 'solid' (as opposed to 'transparent') presentation with more 'body'. (Less ghost-like if you will!!) And horns will definitely not sound laid back! The music will jump at you rather than shyly hiding some place far behind the speakers.

Anyhow, horns have their problems like any kind of speaker, and can be extremely sensitive to the room and the rest of the system. The enthusiast would say that the advantages outweigh the disadvantages and that the problems are challenges.