"The Complete White Paper"

Thank you for your interest in Sound Lab. This paper is designed to introduce you to our technical philosophy, and it is our hope that it will answer any questions that you may have regarding our products.

Sound Lab has been in business since July 1, 1978. The founders of the company, Dr. Roger West and Dr. Dale Ream, shared a belief that electrostatic transduction was potentially the optimum technology for loudspeakers. During this time, the company has successfully addressed the major problems of electrostatic speaker technology and has developed a series of speakers, each of which is designed to fulfill a specific objective. The basic philosophy of Sound Lab is to provide the finest products possible at each of the price points selected. Not only are the products optimum acoustical machines, but they are beautiful pieces of furniture that will grace any decor.

Categorically, Sound Lab is a research and development organization as well as a production facility, as it has elected to develop only truly state-of-the-art products. This implies that the technology base of the company necessarily continues to grow, and this is the case. A strong patent and trademark base has resulted from this continued effort. From the perspective of the customer, however, this could cause some uneasiness since a particular product could possibly become obsolete if the technology level passes their speaker by. To allay this fear, we have established a policy that any product that we manufacture can be brought up to the present technical level at a reasonable cost, or that it may be traded in on a present day product at a generous allowance.

Our products are meticulously handcrafted piece by piece. Each operation is carefully accomplished and checked to make certain that it is correctly manufactured. Batch processing and mass production techniques could not insure the individual quality of each product that we manufacture.

The following section deals with the technical principles upon which our products are based.

1. How Do Electrostatic Speakers Work And What Are Their Advantages?

This section describes the basis of electrostatic speaker technology. The following material is taken from an excellent article written by Jacob Turner entitled: "Why Electrostatics?" The article provides a good tutorial on electrostatic speakers and why they have decided advantages over other speaker technologies.

"Since the latter half of the nineteenth century (circa 1871) the reproduction of sound through electrostatic transducers has stirred the creative vision of professional engineers and idle dreamers alike. It is an interesting fact of history that no other single device in the audio equipment hope chest has enjoyed such an extensive and prolonged courtship between engineer and Audiophile as the electrostatic transducer.

Early attempts to embody this means of sound reproduction were only marginally successful partially because of the lack of suitable materials and processes.

What is the glamour of the electrostatic principle that gained it such extended, devoted attention? Why has the electrostatic transducer remained the standard of excellence by which other acoustic devices are so often measured?

The answer to these questions lies in at least three areas, which will be discussed in the following order:

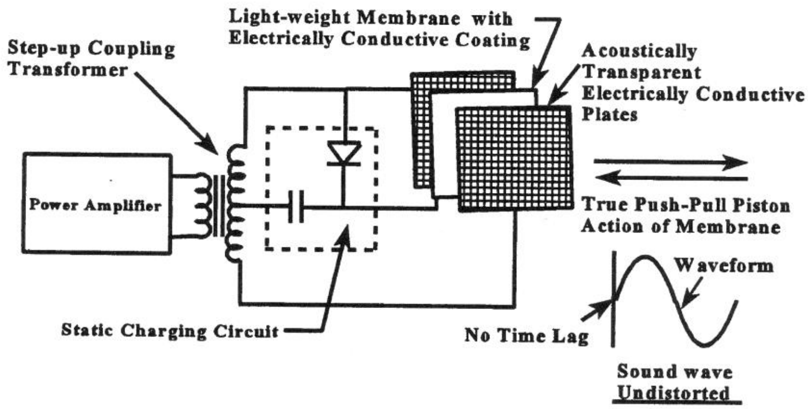
1. Some peculiarities of the hearing process
2. The nature of the acoustic medium: air
3. The operational features of electrostatic acoustic devices as related to the above and to dynamic acoustic transducers

The recent increase of activity in the highly elusive area of psycho-acoustics promises to contribute significantly to a more profound grasp of the complexities of man's perception of his sound environment. Several recent studies have been carried out concerning the sensitivity of normal adults' ears to different levels of harmonic distortion. The results suggest that relatively high levels of harmonic distortion (odd and even order) are imperceptible in the presence of a normal musical program, while quite small changes in amplitude and phasing are readily ascertained. Amplitude changes were described as altering the tonal quality of the program, while phase displacement e.g. bass and treble of no more than 5 degrees degraded the clarity and definition of musical transients and upset the homogeneity of the stereo image between two major frequency bands.

Other studies have pointed out that the inherent transient nature of musical and speech sounds dictates a high level of transient fidelity as a prerequisite of high quality acoustical transducers. The significance of these observations insofar as electrostatic transducers are concerned will be pursued a little later.

Another vital link in the chain is air, which has the following major characteristics of behavior that are germane to our topic. Air is highly compressible, that is, the amount of air pressure (number of air molecules) in a given space can be increased or decreased beyond its normal condition. Air, then, can be said to be like a spring, a means for storing energy; a compliance. Air also has weight or mass. Ten pounds of air are just as heavy as ten pounds of potatoes. Air is, therefore, like an inertance, which opposes an action or force; an inductance. Air can also be randomly excited, air molecules consume power by generating heat. Air can then be said to be like a resistance. This acoustic impedance is normally very low, although at high audio frequencies it is considerably greater than at low audio frequencies. In order to insure that the transfer of diaphragm or cone motion to air motion occurs with the greatest efficiency, it is necessary that the total mechanical impedance of the device be as close to the acoustic impedance of the body of air it is exciting over as much of the audio range as possible.

To relate the preceding discussion to the topic of the electrostatic transducer, it will be necessary to outline the operational features of the push-pull electrostatic device. The previous points will be related at the same time to the operational features of the dynamic transducer. As illustrated in Fig. 1, the electrostatic transducer is composed of a thin membrane (diaphragm) made of Mylar that is stretched and contained between two acoustically open plates. The two plates are connected to either end of a coupling transformer which provides the high voltage audio signal. The diaphragm is connected to a high voltage, low current bias supply, which provides an electrostatic charge that becomes trapped in

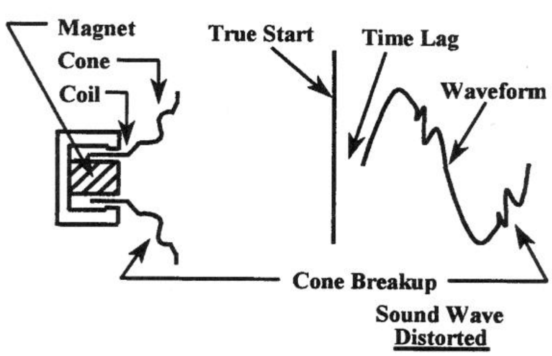


ELECTROSTATIC LOUDSPEAKER

FIGURE 1

the diaphragm. In recent years a method called "electretification" has been developed whereby the bias charge is permanently embedded in the diaphragm material, so that the diaphragm is self-energized without an external source of bias voltage. The net result is the same in both cases. The two plates provide an electric field that is the voltage equivalent of the audio signal. In the presence of an audio signal, the electric field exerts forces on the electrostatic charge that is trapped in the diaphragm. These forces are transferred to the diaphragm, causing the diaphragm to move in synchronization with these forces.

By contrast, Fig. 2 will illustrate the make-up of a dynamic driver, which consists of a frame housing, a magnet, and a voice coil attached to the apex of a cone



CONVENTIONAL CONE LOUDSPEAKER

FIGURE 2

which is suspended at its edge by a flexible cloth or other material. The voice coil is set into motion in synchronization with an audio signal that causes current flow through the coil. As the coil is set into motion by this signal, it in turn sets the cone into motion.

Although both units achieve air excitation through diaphragm or cone motion, the manner in which this is done involves radically different techniques and results. The electrostatic device employs the use of a moving member for all its operating frequencies that is usually only 0.0004 inches thick (Note: the material used in Sound Lab speakers is only 0.00012 inches thick) and weighs only as much as a body of air 7mm thick (about 2mm thick in Sound Lab speakers) whose boundaries are equal to those of the moving diaphragm. The electric field, which acts to make the diaphragm move, exerts its actuating force uniformly over essentially the entire area of the diaphragm. A diaphragm of such extreme lightness, in combination with an actuating force that is uniformly distributed over the entire surface of the diaphragm, results in a transducer whose transient response closely duplicates the electrical input. The net result is a diaphragm motion that is a very good replica of the electrical forces acting upon it, with all sections of the diaphragm surface moving with highly accurate phase and amplitude linearity throughout its entire range of travel, at all frequencies within its area of operation.

The forces acting to move the dynamic transducer's cone, however, produce different results. The application of the driving force only to the apex of the cone necessitates a sufficiently stiff cone to prevent buckling and deformation of the cone structure. Such a stiff cone normally has considerable mass, which degrades its efficiency, its transient response capabilities, and its high frequency performance. In addition, the forces applied at the apex do not act uniformly over the surface of the cone, causing the cone to "break-up" into an infinite variety of vibrational modes, only one of which is truly representative of the original signal. This mode of operation produces amplitude and phase nonlinearity often of considerable magnitude, and these tend to increase as the cone is driven to greater excursions. Obviously, the discussion of dynamic driver operation relates quite strongly to the previous discussion concerning the unusual sensitivity of the human ear to the problems of transient response, amplitude linearity, and phase linearity. The basic conclusion is that an electrostatic unit behaves with better composure in all of the above areas.

The second major distinction involving electrostatic transducers deals with the considerable problem of coupling to the air with reasonable efficiency over the entire audio band. The electrostatic unit, because of its extremely low mass diaphragm and the uniform distribution of the driving forces over the entire diaphragm surface, is inherently a unit with low mechanical impedance at all frequencies. As such, the air-coupling problem at low frequencies (where the problem is greatest) for electrostatic units is considerably less than for dynamic units, which are encumbered by a high mechanical impedance. The result of these conditions is that the electrostatic unit performs quite well down to its frequency limits and within its maximum excursibility with equal fidelity at all drive levels.

The dynamic cone unit, because of its poorer coupling, must be driven harder to produce satisfactory excitation of the air at low frequencies, and usually encounters a number of problems involving cone break-up, non-linear motion of the voice coil due to loss of magnetic coupling in the gap, suspension non-linearities, etc. In all fairness, it must be said that the performance level of today's popular dynamic acoustic transducer is incredibly good given the economic and operational constraints of that type of unit. On the other hand, the superiority of the electrostatic principle has been demonstrated by the great acceptance of the increasing number of electrostatic headphones which have already emerged in the market. In addition, of course, several electrostatic speaker products are highly regarded by the audiophile community."

We greatly appreciate Jacob Turner's ability to explain the advantages of electrostatic technology over conventional dynamic (magnetic) technology. We would add that the comments regarding cone speakers apply to ribbon and other types of speakers also. In the case of ribbon speakers, they are planar in nature but the percentage of the area of the diaphragm that is directly driven is on the order of 30% to 50% instead of 100% as in the case of electrostatic speakers since the driving force comes from a thin piece of wire or foil that represents only a fraction of the total moving area. Thus, as in the case of the cone speaker, a significant portion of the diaphragm is not directly driven and can inherit some of the drawbacks of cones. Furthermore, the wire or ribbon is usually sandwiched between two plastic films which adds to the moving mass. In general, the ribbon speaker is better than its cone counterpart in terms of motional accuracy, but it has upwards of ten times the moving mass of a properly designed electrostatic speaker.

2. What Is The Relationship Between Speakers And Microphones And How Is It Optimized?

The goal of high-end audio is to recreate a sonic performance with as much accuracy as possible. This demands near-perfect performance from every component in the reproduction system, including the sound room. But it also requires a compatibility with certain circumstances involved in the recording process which have to do with the accuracy of the recreated stage (stereo image, both size and location), ie: the spatial aspect of the re-creation. Perhaps, the more important of these has to do with the characteristics of the microphone(s) used to record the performance and their placement.

Not much can be done to compensate for signal timing corruption due to mix-down procedures and bizarre microphone techniques. These are in the hands of the recording engineer. But, something can be done concerning an acoustical principle that we refer to as microphone/speaker reciprocity. In the early days of stereophonic recording it was generally agreed that the “standard" microphone arrangement was to have two microphones facing the orchestra which were generally separated by eight feet. There have been many alterations to this simple geometry since then, such as employing more than two microphones to obtain special effects and sonic highlighting. However, amid the rather complex science of recording, one generality persists. Most microphones used in recording employ a cardioid (directional) pattern. A microphone having this characteristic readily accepts information from the front but attenuates sound waves approaching from the rear. On the average, the cardioid acceptance pattern is such that pickup sensitivity is diminished by 6dB (half pressure) at about plus or minus 45 degrees off of the central axis of the microphone.

The principle of reciprocity simply states that in order to recreate a sound stage with good spatial accuracy, the dispersion angle of the loudspeaker and the acceptance angle of the microphone should be similar. For example, were you to walk around your sound room and compare the recreated sound field to walking around the auditorium during the actual performance, similarity of the stage image can only be achieved if the principle of reciprocity is implemented. For this reason all Sound Lab speakers are carefully designed to fulfill this important principle. More about the requirements for accurate staging will be discussed in a later section.

Another advantage that is accrued from a wider dispersion angle is that harsh direct reflections from walls are greatly reduced, making speaker setup much easier since undesirable room modes tend to be less pronounced. Through experience we have found that horizontal dispersion angles from 45° to 90° work well in the home environment. The more narrow angle provides greater speaker sensitivity and dynamic range while the wider angle provides a wider stage. Both angles are available in our products.

3. Sound Lab Speakers Don't Seem To Sound Any Louder Up Close Than They Do Far Away! How Is This Done?

The secret behind this seemingly magical trait has to do with the characteristics of the acoustical line source. All Sound Lab products are categorically vertical acoustical line sources. My opinion is that the vertical line source is the optimum acoustical geometry for proper staging and acoustical energy control in the home listening environment, which I will now attempt to justify.

Theoretically speaking, a vertical line source is a long, one-dimensional vertical line (having length but zero diameter) from which sound energy emanates. It might be helpful to envision a violin string, but much longer in order to produce lower frequencies. Furthermore, the length of this line needs to be large compared to all the wave lengths that it transmits. There is an important exception when used in an enclosed room, as will be discussed later. If this criterion is met, the major characteristic of the line source is that it disperses energy only in the horizontal direction and none in the vertical direction. In other words, all sound rays emanate perpendicularly to the vertical line. Therefore, the radiation field might be compared to the shape of a slice of cheese cake in which the top and bottom are flat since there is no vertical radiation. Most speakers available today are technically "point sources". The theoretical characteristic of a point source is that the source of sound has zero dimension and it uniformly radiates energy in all directions: up, down and sideways, etc. In the real world a point source is a speaker that is dimensionally small compared to all of the wavelengths it reproduces.

My justification of why I believe the line source to be superior to the point source in the home environment will now be answered. For the reader who doesn't wish to get entangled in mathematics, I will first give an explanation that appeals to the intuition. This will be followed by a more rigorous mathematical approach.

It is intuitively apparent that if the infinitely long vertical line source radiates at perpendiculars to the line, there would be no ceiling and floor reflections to cause wave interference patterns. In other words, the sound would not be as cluttered with delayed replicas of the direct sound as is the case of the point source, which radiates vertically as well as horizontally resulting in greater wave interference activity. Wave interference, caused by the interaction of reflections with themselves and the direct sound from the speaker, corrupts the frequency and phase response of the sound.

In order to obtain the wonderful characteristics of the line source over the full audio spectrum, it theoretically requires a line source a length several times the wavelength of the lowest frequency of interest. For example, if 20 Hz is the lowest frequency of interest, the length of an adequate line source should be on the order of, say, ten times the wave length, which would give a line length approaching 550 feet! Not very practical for a listening room. Otherwise, the rays will have a vertical component to their path, which will introduce vertical reflections. For a line source of finite length, this effect becomes more pronounced as the wavelength increases (ie: as the frequency is lowered). This immediately appears to rule out realizing the benefits of an adequately long line source in the home.

Here's the secret on how to fit a very long line source in to a normal-size listening room. Reflections from the ceiling and floor of the listening room, which are due to using a truncated line source, actually extend the virtual length of the line source infinitely in both the upper and lower directions. This sounds incredible, but here's the reason. To demonstrate this, imagine an infinitely long line source that just fortunately happens to pierce your ceiling and floor exactly in the location that you would want to place your speaker, and this line just happens to be perpendicular to the floor and ceiling. If the ceiling and floor are sufficiently reflective (a condition, incidentally, that is undesirable for the point source where it is desired to absorb energy directed at the floor and ceiling) it can be shown mathematically that due to the presence of the floor and ceiling, the line above the ceiling and below the floor are "phantomized" (reconstructed from reflections). That is, that portion of the infinite line source that occupies the room still behaves as if it is infinite in length. Thus, due to reflections from the floor and ceiling, the truncated line source becomes a virtual infinitely long line.

Looking at it from another direction, since the sound rays emanate at perpendiculars from the infinite line, introducing a floor and ceiling should have no effect on the characteristics of the line between the floor and ceiling since they are parallel with the sound rays and therefore don't interfere with them. Theoretically, the only requirements are that the floor and ceiling are perpendicular to the line source and that they are sufficiently reflective. In the real world, the floor and ceiling will absorb some sound energy, but the effect is not sufficient to alter the line characteristics enough to be audible. The net result is that the listener can sit down, stand up, do deep-knee bends, walk back and forth and the sound image does not appear to change. A point source cannot give this kind of performance. Another benefit of the line source is that the effects of room modes are not as apparent since the contribution of vertical reflections is eliminated.

Finally, since the line source has no vertical dispersion, all energy is concentrated in the horizontal direction. The theoretical effect of this, ignoring the reflective field resulting from wall reflections, is that sound drops off at only 3dB per doubling of distance from the speaker rather than 6dB per doubling of distance as in the case of the point source. Therefore, as the listener walks toward or away from a line source, changes in loudness are much less apparent. That is why the sound level of our speakers does not seem to drop as one walks away from them. Conversely, it is why the sound level of our speakers does not appear to become louder as one approaches them.

To make this more meaningful, I shall be somewhat rigorous mathematically. The acoustic intensity (I) of a sound wave is defined as the average power transmitted per unit area in the direction of wave propagation:

$$I = \frac{P_{rms}^2}{\rho \times c}$$

Where:

I = average power transmitted per unit area in the direction of wave propagation

P_{rms} = effective pressure in nt/m² (newtons/square meter)

c = velocity of sound in m/sec (meters per second)

ρ = density of air in k/m² (kilograms per square meter)

With the point source, the area that energy is distributed over is quadrupled each time the distance from the source is doubled. Thus, in decibels, the intensity of a sound (acoustical energy density) drops approximately 6.02dB for each doubling of distance:

$$db = 10 * \log_{10}\left(\frac{I_1}{I_2}\right) = 20 * \log_{10}\left(\frac{p_1}{p_2}\right)$$

Since doubling distance reduces sound intensity (I) by a factor of four:

$$\Delta SPL = 10 * \log_{10}\left(\frac{1}{4}\right) = -6.02dB$$

In comparison, with the line source area, the sound level changes only by a factor of two for doubling the distance toward the line source, since doubling the distance distributes the sound energy over only twice the area. Thus, doubling the distance from a line source reduces the sound level by only 3.01 dB. In other words, sound intensity drops off with distance significantly slower with the line source than with the point source.

I'd like to now show that beyond a certain distance, the sound level emanating from a line source will be louder than that of a point source. You may have already surmised this since the output of a line source falls off in magnitude at a slower rate than that of the point source. The interest here is to be able to calculate what this critical distance is, where the output of the line source and that of the point source becomes equal, and beyond that point the line source will always be louder.

When comparing standard sensitivity measurements of point sources versus line sources (usually one watt applied at the input of the speaker and the resulting sound level is measured one meter from the speaker), it is common to misinterpret the results as it applies to sound pressure levels at standard measurement distances. The reason for this controversy is due to the proximity effect of the line source compared to that of the point source. Sound level varies as the inverse of distance for the line source, whereas it varies as the inverse of the **square** of the distance for the point source.

As a result of this difference in proximity characteristics, comparing the standard sensitivity measurement values of the line source and point source to the sound pressure level measured at your favorite listening location rather than at one meter, the result can be startling. Generally, the point source speaker has a higher energy density at the standard one-meter distance and therefore usually has a higher standard sensitivity rating than that of the line source. However, when the sound level is measured at normal listening distances, this difference in sound levels becomes less due to the differing proximity characteristics of the line and point sources. Since this advantage increases with distance, there comes a point, we term as the "critical distance" where the line source equals the sound level of the point source. Beyond this point, the line source continues to become louder than the point source as distance is further increased.

Thus, when standard one meter sensitivity measurements of a line source and a point source are compared (defined as the sound-pressure-level resulting from one watt of power applied at the speaker terminals and measured at a distance of one meter) the results can be deceiving. It only makes sense to measure sensitivity at a "normal" listening distance rather than at one meter when comparing point-source and line-source speaker sensitivities.

Perhaps, the best way to relate standard one-meter sensitivity values with the performance at typical listening distances is to calculate the distance from the speaker at which the sound level is the same as a function of the difference of the one-meter sensitivity ratings of the two speakers. Keep in mind that since the line source sound level drops off slower than the point source, there inevitably is a distance from the source where the sound level will be the same, beyond which the line source will always be louder than the point source. In fact, the degree that the line source is louder than the point source is proportional to this distance. Anechoic conditions are assumed.

The formula for this critical distance (D), where the line source and point source have the same sound level, is given by:

$$\mathbf{D = 3.28 \times (2^{[(S_1 - S_2) \div 3.01]} + 1)}\ \boldsymbol{ft}.$$

Where:

D = The distance from the speaker, in feet, where the line source and point source speakers have the same sound level

$\mathbf{S_1}$ = One-meter sensitivity of a point-source speaker (dB/1W/1M)

$\mathbf{S_2}$ = One-meter sensitivity of a line-source speaker (dB/1W/1M)

S_1 and S_2 are obtained using the standard method of measuring speaker sensitivity by applying 1 watt of signal energy at the input terminals of the speaker and measuring the SPL (in dB) at a distance of one meter from the speaker on axis (for an 8 ohm speaker this would be 2.83 VRMS applied at the input terminals). This equation assumes that the point-source speaker sensitivity (S_1) is greater than that of the line-source speaker near the speakers, which is usually a safe assumption.

This equation is derived from the fact that the line-source obtains a 3.01 dB advantage over the point source with each doubling of distance from the source.

For example, our Model A-3PX speaker has a one-meter sensitivity rating of about 88 dB. Let's compare that to a hypothetical "Brand X" speaker having a high sensitivity of 96 dB. The critical distance calculates to be about 24 feet. Beyond this distance, the A-3PX will be louder for the same input voltage. Acoustical environments that are only partially absorptive, such as the average home listening room, will cause some variation in actual measurements, but the general effect will be the same. The moral of the story is that to properly compare the useful sensitivities of a line-source speaker versus a point-source speaker, measurements should be taken at normal listening distances rather than at the standard 1 meter.

Now, from a pragmatic viewpoint, let's interpret the above conclusions in terms of advantages in the listening room. First, imagine the sound of a live orchestra in a symphony hall. As one stands near a side wall, the closer musical instruments appear louder than the ones further away, but they will not mask (override) them. As one walks to the other side of the hall, a similar effect is observed. In other words, the acoustical sound stage is skewed as one stands to the side, but a three-dimensional image of the entire orchestra remains since the closer instruments do not mask the more remote ones. This effect is termed “staging” and is very similar to the effect of listening to music reproduced through stereophonic line sources. That is, as a listener walks to one side of the listening room the speaker nearest him does not become significantly louder and, hence, does not mask the sound of the other speaker. In fact, as the listener moves around the listening room the spatial image of the reproduced orchestra responds very similarly to that of an actual orchestra playing in a music hall. In contrast, the point source speaker becomes significantly louder as one approaches it and is capable of masking the sound of the more distant speaker, which represents poor staging. A point source speaker requires that a listener remain close to the center line between speakers to receive proper orchestral balance. In other words, good staging is inherent in line-source speakers.

Since the sound rays emanating from a line source depart perpendicularly from the line, the sound that reaches the ear comes from the line-source at the same height as the ear. Hence, as the listener stoops or stands, the sound that is heard comes from the same level as the listener's ear. Thus, one never listens "down to" or "up to" the sound source. In contrast, the point- source has a very distinct vertical position, and can be easily localized when doing deep-knee bends. This may give music an artificial vertical localization effect. Contrary to some common beliefs, stereophonic music does not contain any vertical localization cues and, therefore, a true vertical image does not exist. To create a true vertical image it would require an upper and a lower set of speakers (with the corresponding four-channel microphone/recording setup with which to make such a recording) to simultaneously reproduce the vertical and horizontal images. Thus, the line-source speaker does not create artificial vertical localization such as multiple driver point-source speakers can do.

4. How Does Sound Lab Achieve Profound Bass Response With Dipole Panel Speakers?

Sound Lab products employ a revolutionary patented principle, called "Distributed Bass Resonance", that virtually eliminates the two major drawbacks of dipole speakers: the membrane "drum-head" resonance and dipole energy cancellation. These will now be explained and how their effects are eliminated by Sound Lab's technology.

A resonance frequency is exhibited by all taut membranes due to their mass and compliance. This is an important property of some things such as a drum, but it can be a serious drawback in the electrostatic speaker (or any type of membrane speaker) if it is not eliminated. It can cause a large acoustical energy peak at the resonant frequency that can be as high as 30 dB. This sonic peak not only "colors" the sound but it also limits the usable dynamic range of the speaker.

A common approach used by some designers to eliminate the resonance peak is to employ an acoustical damping material that the sound must pass through, such as a fine mesh, similar to that used in the manufacture of sheer nylon stockings. Unfortunately, resistive damping does "surgery" over the entire pass band of the speaker, giving it a lackluster quality. Using this

approach the resonant energy is dissipated and not put to good use.
We considered the possibility of using the membrane's resonant energy constructively rather than dissipating it, since one characteristic of an under-damped membrane at its resonant frequency is that it is very responsive. In fact, it can be so responsive that if it isn't controlled it can cause the diaphragm to slap the stator electrodes at relatively low signal input levels. The reason why this effect is so dramatic is because every unit of diaphragm area to the peak of energy at resonance.

A philosopher once stated: “Asking the proper question leads to the answer". We asked: “instead of permitting the entire diaphragm to contribute to one resonant peak, why not set up a situation where different sections of the diaphragm resonate at different frequencies in a graded fashion such that the resulting set of resonant frequencies distributes resonant energy over the lower frequencies?” Two wonderful results are realized: the "drum-head" resonance is eliminated and the efficiency of lower frequencies is dramatically increased. The overall result is bass response that is devoid of single-peaked (juke-box) bass and, even more important, it is fast, dynamic and unrestrained.

The distributed resonance principle also solves a nasty problem associated with dipole radiators: dipole cancellation. An acoustic dipole radiator is basically a vibrating membrane in which the acoustic energy emanating from both of its sides is permitted to propagate freely. Dipole radiation preserves the purity of the sound since it doesn’t use enclosures, which can “color” the sound due to internal resonances. The problem arises from the fact that an acoustic wave radiates from each side of the diaphragm, both of which are mutually out of phase. Since they have opposite phase they begin to cancel one another at lower frequencies, which weakens bass energy. By judiciously selecting the “law" by which the resonant energy is distributed, the effects of dipole cancellation can be eliminated. Figure 3 shows the typical response of an un-equalized dipole radiator taken axial to one side.

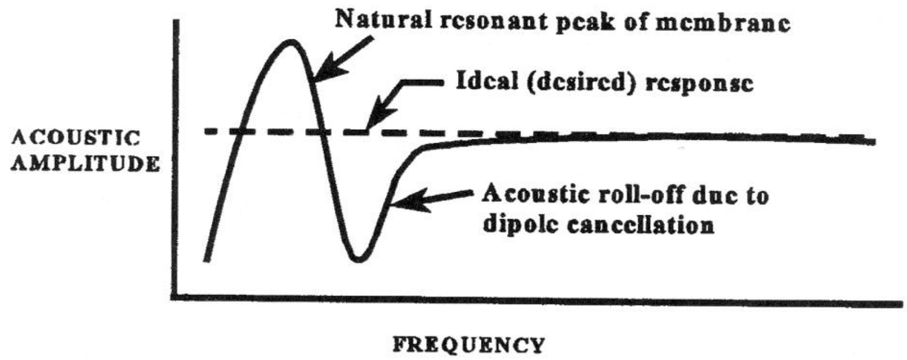


FIGURE 3: Typical "saddle" response of an unequalized dipole diaphragm

Figure 4 shows one of several methods of sectoring a diaphragm to distribute resonant energy. f1, f2, etc., represent the resonant frequencies of each of the sectors.

FIGURE 4: Membrane is shown divided into independent sectors. f_1, f_2, etc., represent the resonant frequency of each sector.

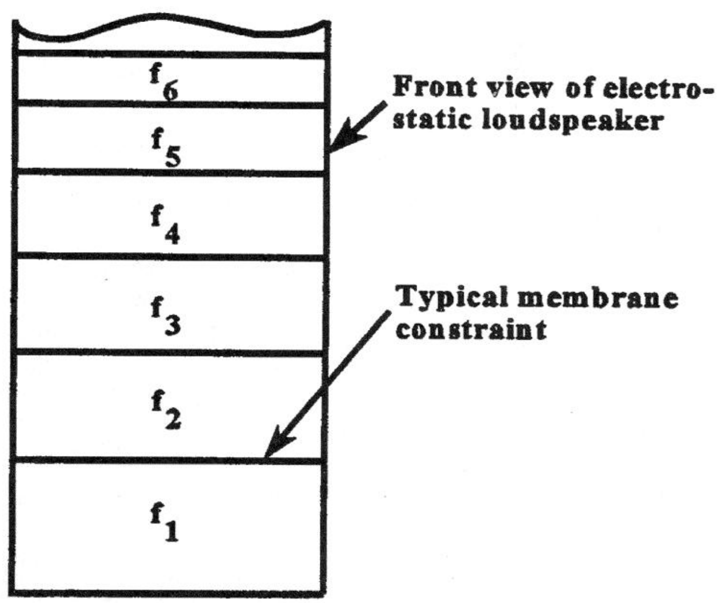


Figure 5 shows the resulting flat acoustic response.

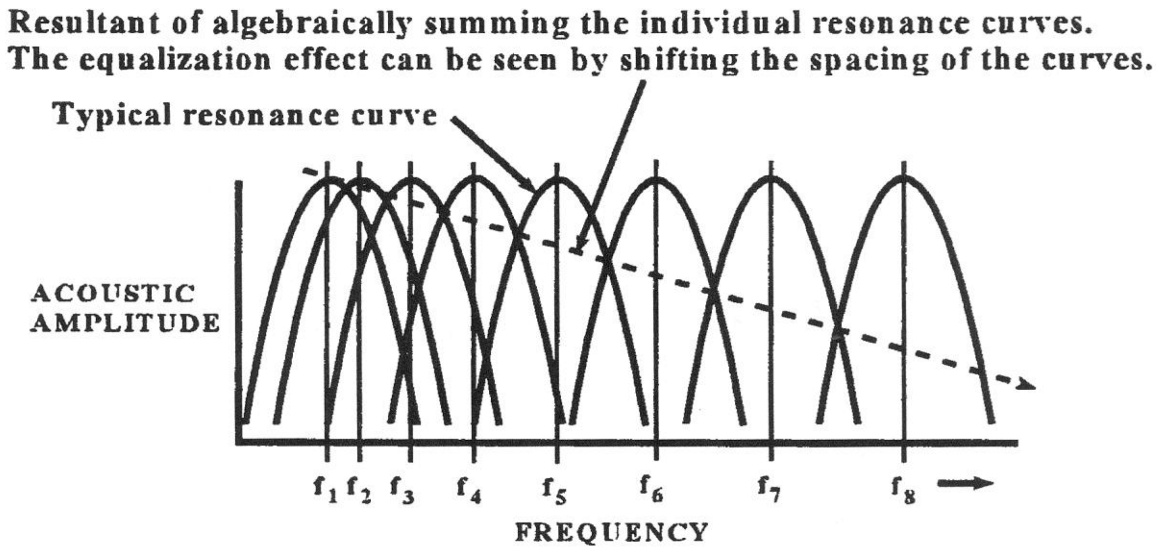


FIGURE 5: **Flat low-frequency response obtained by summing distributed resonance energy**

In the “near-field" (that is, very close to the speaker) the bass response of the speaker has a rising effect as frequency is decreased. This occurs because up close to the speaker the effects of dipole cancellation are not as apparent. However, in the far-field (that is, normal listening distances and beyond) the rising characteristic seen at the "near-field" adds just enough energy at each frequency to make up for the energy lost due to dipole cancellation.

The resulting frequency response is flat. In other words, the “law" of the distribution of frequencies is such that it is the complement of the speaker's frequency response curve when it is under the influence of dipole cancellation. Obviously, an electronic equalizer could be used to obtain the same effect, but it would reduce the effective low-frequency dynamic range of a given amplifier on the order of 15 dB or more. By using "distributed-resonance equalization" the dynamic range of a given system is not compromised and it eliminates the requirement of an electronic equalizer. This is about as close as one can come to "having your cake while keeping it".

5. Electrical Charge: The Backbone of the Electrostatic Speaker:

In a conventional (dynamic) loudspeaker magnets are used to create a force field. Electrical charge is employed to do the same in the case of the electrostatic speaker. The quality of the performance of the electrostatic speaker is directly associated with how uniform and dense the charge-field is distributed on the membrane. Sound Lab speakers employ a pure copper charge diffusion ring around the periphery of the membrane to feed charge into the interior of the membrane from all directions. The low electrical impedance of this ring insures that leakage paths, such as moisture condensing onto the membrane as a result of high humidity, has no effect on the performance of the speaker. The resulting charge field is exceptionally uniform and dense over the entire membrane's surface.

Furthermore, since the charge is fed from all points around the periphery of the membrane, the speaker "charges up" virtually instantaneously. Therefore, the speaker can be unplugged from the mains if desired without requiring a re-charging period.

6. Geometrical and Construction Considerations:

The unique geometry of our electrostatic panels, along with adherence to close tolerances, are the keys to the accurate three-dimensional staging of our speakers. First, the spacing between the stators and the membrane is critical in order to obtain the proper electrostatic field intensity and to have adequate room for large membrane excursions. Further, the spacing from the stators to the membrane must be closely the same for both the front and back stators to have optimum push-pull action and cancellation of distortion products.

The framework of the panel is also critical. The whole purpose of the framework is to maintain the proper geometrical relationship between the stators and the membrane. The panel frame must be very rigid, accurately constructed and non-resonant. Furthermore, the material used must be moisture resistant and not degrade with time and environmental conditions. We therefore fabricate the internal members of the structure from a plastic that is strong, dense, non-resonant and non-hygroscopic. The integrity of the panel must be preserved under all reasonable conditions.

Some Criticisms of Electrostatic Speakers that no Longer Apply

Perhaps, the greatest general criticism directed at electrostatic speakers is that they easily "arc over" electrically (ie: break down) and are thereby easily damaged. A breakdown is the result of a failure in the stator insulation, which results in an electrical arc jumping between the membrane and the stator. Because the excellent insulating materials of today were not available in the earlier days of electrostatic speaker design, breakdown was relatively common, which created a bad name for electrostatic speakers. Understandably, many audiophiles tended to shy away from them. When a breakdown occurred, it usually burned holes in the membrane, announcing the situation with loud popping sounds.

Sound Lab has continually researched insulation materials. Recently, this effort resulted in the discovery of an insulation that can withstand the tremendous electrostatic field intensities required to reproduce sounds at realistic sound levels without failing. A Sound Lab panel will never break down, even under overload conditions, which eliminates the greatest concern of the owner of our speakers. To insure the integrity of each speaker we manufacture, it is tested beyond the rated maximum power before it is shipped.

Another criticism of electrostatic speakers, again based on early designs, is that they have limited horizontal dispersion, which results in a limited "sweet spot", whereby the listener must sit in a particular location in order to hear the best sound from the speaker. We, at Sound Lab, have adopted a piece-wise approximation of a curved surface to disperse the sound. This approach permits dispersing the sound to any desired degree, ranging from negative angles (focusing) up to a full 360°, without spatial anomalies such as the "picket fence" effect or off-angle coloration (spectral characteristics changing with listening angle). The wave front is smooth and the spectrum of the sound remains unchanged over the full angle of dispersion. The result is a "sweet spot" the size of the listening room. This is accomplished by placing a series of narrow flat panels (facets) side-by-side such that they approximate a curved surface. The angle between the facets is critically adjusted so that the sound patterns of adjacent facets at frequencies as high as 20KHz overlap such that there is no acoustical dead-zone between them. Furthermore, the width of the flat facets is made sufficient to permit the production of bass frequencies. This results in a smooth wave-front over the entire audio spectrum, and is the basis of our full-range electrostatic panels. The use of just one piece-wise curved membrane to reproduce the entire audio spectrum provides perfect time alignment and smooth, controllable dispersion.

Other designers have adopted other methods to disperse sound energy, each resulting in a limitation. One approach uses a narrow strip panel for the higher frequencies. This is based on the natural law of energy dispersion whereby as the wavelength of the sound is large compared to the width of the radiator, energy is dispersed. Unfortunately, if the strip is made small compared to the higher audio frequencies, the width of the speaker would have to be about half an inch wide for good dispersion, which is too narrow for acceptable energy production at lower frequencies. Necessarily making the width larger results in a graded dispersion whereby lower frequencies disperse more than higher frequencies, resulting in a limited "sweet spot".

Another dispersion approach employs a membrane that is truly curved in the horizontal plane. With a bit of thought it becomes obvious that such a geometry provides a non-symmetrical transduction characteristic since the membrane tension increases as the membrane expands outwards and it loses tension as it moves inwards. This non-symmetrical characteristic creates audible distortion on larger excursions of the membrane, especially at lower frequencies where excursions are greater. This dispersion approach clearly cannot be used for a truly full-range speaker. In fact, the greater the angle of curvature used, the more severe the limitations become.

In brief, Sound Lab's approach, using narrow flat panels in a piecewise approximation of a curved surface, provides the basis of a single-membrane, full-range speaker having smooth wide horizontal dispersion, perfect time alignment, low distortion and excellent dynamic range.

The membrane is another critical part of the speaker panel and should at least be briefly mentioned. The membrane must have low mass and excellent resistance to environmental factors. Sound Lab uses a polyester material that has a thickness on the order of only 100 millionths of an inch thick. This provides excellent compliance for large, linear membrane excursions. Also, the moving mass is so low that it represents only a fraction of the mass of the air volume it moves. Furthermore, Sound Lab uses a unique mechanical/adhesive system to insure that the membrane never slips which would cause it to lose its proper tension. Finally, the membrane is coated with a very thin proprietary substance to give it the correct electrical conductivity in order to hold a dense charge that's uniform over the entire membrane surface. This coating is virtually indestructible and is not affected by normal environmental factors.

In 2006 Sound Lab introduced an advanced electrostatic transducer design approach in order to meet the high standards of professional audio. A series of speakers has evolved from this effort. Since then, this new technology has been implemented into our high-end audiophile speakers. A paper has been attached that discusses this new technology.

Congratulations! It understandably requires reasonable effort to work through the details of this paper. Much more could be presented to explore our technology in more detail, but this covers the more important areas. The real test is in the sound. We are proud of our technology, and we feel that our speakers are based on the most important acoustical principles that apply to sound reproduction. Please take the time to make a critical comparison of our products with other brands. You will be glad that you did!

If this presentation has not answered all of your questions, we invite you to contact us for the answers.

We also welcome any suggestions that you may have.

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Differences Between Sound Lab's new PX Technology and the Previous Technology

There have been numerous inquiries about the difference between our older panel technology and our new PX technology. There are quite a few improvements, but for the sake of brevity only the major ones will be mentioned.

To assist understanding of the improvements, a brief explanation of the essential aspects of the electrostatic panel is now given. The electrostatic panel consists of two main functional parts, the stators and the membranes. A dense (high-voltage) static charge is applied to the membrane and the audio signal, stepped up in amplitude to be approximately 100 times greater than the amplifier's output signal, is applied to the stators. The resulting electrostatic forces cause the membrane to move in accordance with the signal. The metallic stators require electrical insulation to prevent electrical arcing between the membrane and stators. Furthermore, the membrane needs to have a very low mass along with a low-mass electrically- conductive coating in order to hold an electrical charge. Essentially, this describes an electrostatic panel. A more detailed explanation is provided in the technical section of our website:

www.soundlab-speakers.com.

From a practical standpoint there needs to be open space in the stators for the sound to pass through. We have determined that about a 40% acoustical openness in the stators is optimum for maximum acoustic efficiency. The stators consist of a planar grid made of insulated wires. The more wires included in the stator, the higher the coupling efficiency to the charge on the membrane. The number of wires that can be used, for 40% openness, is controlled ultimately by the thickness of the insulation on the stator wires. The number of wires on the stators of the previous technology was less than that of PX technology because the insulation had to be thicker. Through our research we discovered an insulation polymer that can be significantly thinner without compromising voltage-handling capability. In fact, the new polymer permitted the stators to have nearly a 50% increase in wire density in addition to significantly greater reliability, as the new polymer is much tougher both electrically and mechanically. Basically, we have had no failures due to electrical breakdown since we introduced it. This improvement has increased the efficiency of the panel from about 6dB to 9dB, depending on the vintage of the speaker.

The membrane coating has also has been improved. The new coating is much more durable and is very resistant to environmental factors, such as humidity. We have units everywhere in the world and have had no problems, even with high humidity near oceans where salt content is high. In addition, we have developed a superior means of securing the membrane to maintain the optimum tension, permanently.

The panel framework of the new PX panels is much stronger and stiffer than the older framework. We now use mechanically interlocking parts in conjunction with adhesives and fasteners. The resulting framework is free of all audible mechanical resonance. Furthermore, it provides the necessary close tolerances. Demands for greater efficiency and dynamic range stimulated our development of PX technology. We have introduced several commercial speakers using this technology and the results have been quite excellent. As you can imagine, there's no room for failure in a professional installation, and efficiency and loudness have to be adequate to compete favorably with standard dynamic technology in the industry.

We have also reduced the curvature of the panels to 45 degrees rather than 90 degrees used in our older panels. Theoretically, this doubles the energy density, which increases the efficiency and dynamic range of the speakers by another 6dB. Since reflections in most sound rooms add to the perceived loudness, reducing the angle to 45 degrees gives about 3dB enhanced efficiency. The 45 degree horizontal dispersion angle has been quite well received. Not only does it increase sensitivity and dynamic range, but it also reduces wall reflections. Many listeners claim that the imaging is more detailed, which more than likely results from less wall-reflected energy. The result is greater clarity, impact and stunning imaging.

The backplates (electronics) have not been ignored. We have introduced an option that we refer to as the "hot-rod" update. This update uses the highest quality components available at any cost without being impractical. The result is improved low-level resolution and more “slam” in powerful sonic wave fronts. This is due mainly to the employment of special capacitors that have highly conductive foil electrodes in conjunction with an oil dielectric. These capacitors are capable of large current transfer and have virtually no measurable dielectric absorption. The latter is a measure of energy that a capacitor stores and doesn't release quickly. It causes a form of distortion that reduces clarity. With the oil-filled foil capacitors, a decaying musical note can be heard down to the threshold of hearing due to the enhanced low-level resolution. It should be mentioned that our standard high-quality polypropylene film capacitors have excellent characteristics, and satisfy the majority of listeners, but to the few who won’t compromise, the foil capacitors and super-stable resistors used in the Hot-Rod version are indispensible.

The net result of PX technology is an increase of efficiency of up to 9dB over our older technology. Furthermore, audible structural resonance is eliminated, reliability has been virtually perfect and, due to the wider dynamic range, the speakers have a greater sonic impact and dynamic range. Bass is fast, robust and detailed. Mids have a “I’m there” presence. Highs are extended, detailed and silky. All of this from a single membrane, which provides perfect time alignment, and is the secret of the remarkable imaging and staging.