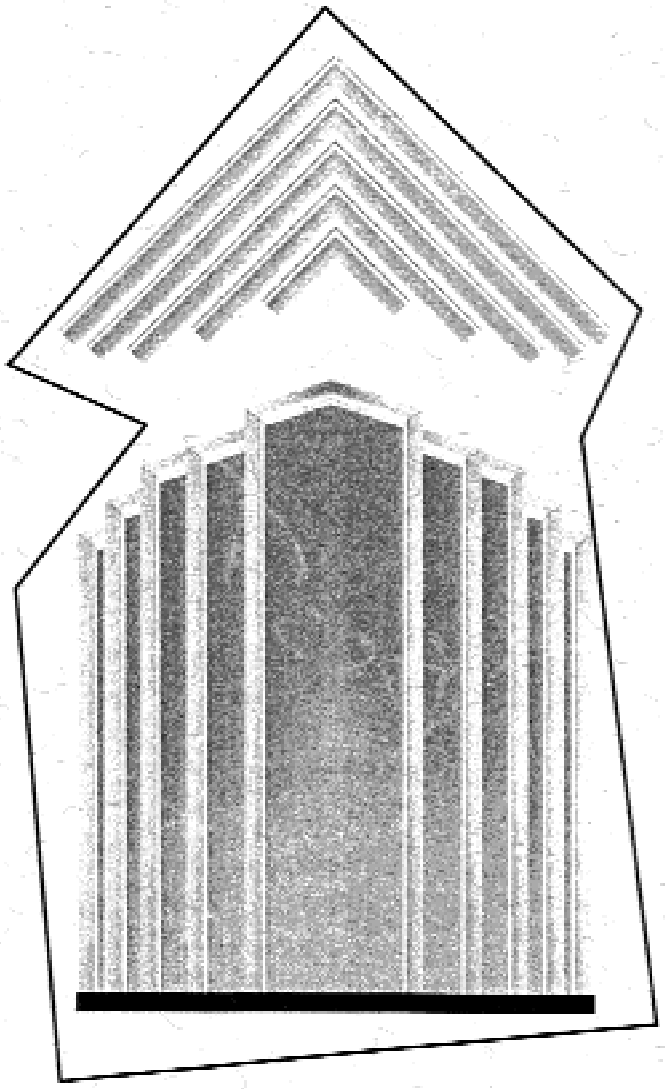
SHOCK WAVE

ELECTROSTATIC

SUBWOOFER



DAVID LUCAS

ELECTROSTATIC MAGIC IN THE CLASSIC TRADITION

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Best wishes

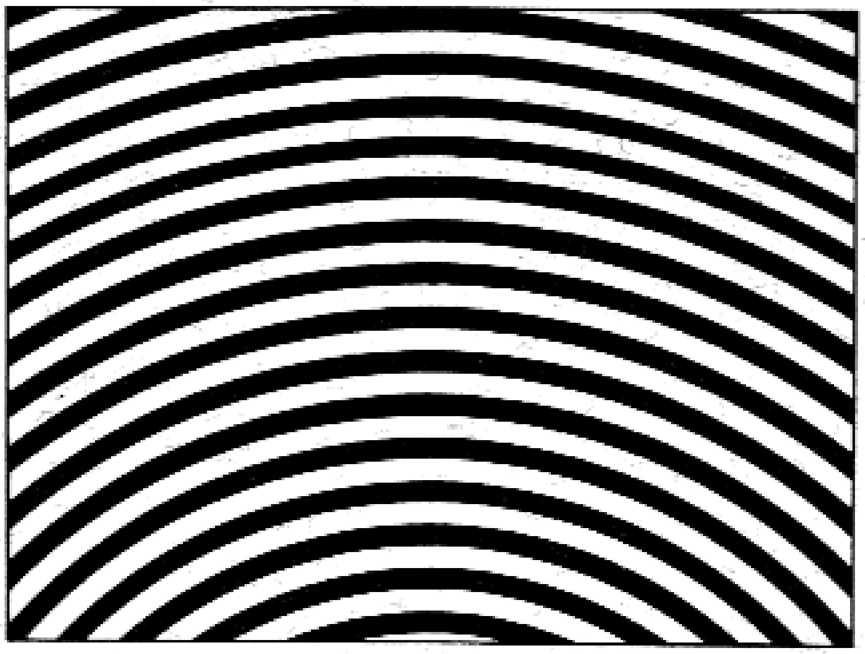
DAVID LUCAS

SPECIAL NOTES

1. This manuscript was written with the assumption that you have a general or basic working knowledge of electrostatics, high voltage, and electronics. We assume no responsibility for the use or misuse of our products. In particular, it uses VERY HIGH VOLTAGES that must be treated with respect. Voltage this high likes to "reach out and touch someone"; It likes to break some of the basic rules you may be familiar with and behave by a new set of rules you may be UNfamiliar with! So practice all the safety rules that apply to high voltage and be sure to ALWAYS use protective grills and/or grill cloths to keep out and protect "curious hands"!
2. I have deliberately left out ALL of the equations and mathematics that I have used to develop and create SHOCKWAVE. Those of you that seek more academic knowledge can refer to the chapter entitled "References".
3. Many security measures have been used in this manuscript for obvious reasons. This idea protects YOUR investment as well as mine. ALL of the construction details are here --- but they may take the form of models and samples.
4. The general construction is very inexpensive and simple since you only build FLAT electrostatic loudspeakers(ESLs). Since there are so many schools of thought on what is considered the superior ESL construction method, I am unbiased and have deliberately left out these basic construction methods to let you decide on the one that best suits your needs. Audio Amateur Corp. (603-924-6371) has many past articles and books that address basic ESL construction.

BASS BASICS

The definition of ideal low-frequency sound reproduction is actually fairly simple (it is the realization that poses the difficulties). Although the definition is simple, it may not be universally accepted by all designers, manufacturers or even listeners. The adopted definition is precisely the same one that should be applied to a full-range sound reproducer, i.e., "The sound pressure at the listener's ears must exactly follow the desired input electrical waveform." Any deviation in that sound pressure waveform is a distortion. In practice, it is impossible to exactly duplicate any input waveform in this manner. But advances in our understanding and the creation and usage of new materials can, and have, progressively closed the gap and reduced distortions. However, to successfully achieve a significant advancement, perfection must be the goal rather than merely attempting to create "something different" or "something better". To produce a product which performs its acoustic function in a nearly perfect manner, a designer must decide which performance characteristics can withstand compromise and by how much—and which cannot. The only allowable reason for not maintaining perfection in all areas of performance must be that the human hearing mechanism is unable to detect errors in this characteristic. No consideration can be given to what is possible or feasible to produce. To do so would lead to a less than perfect product, one that would naturally perform very much like past and present products. The list of ideal performance characteristics are:



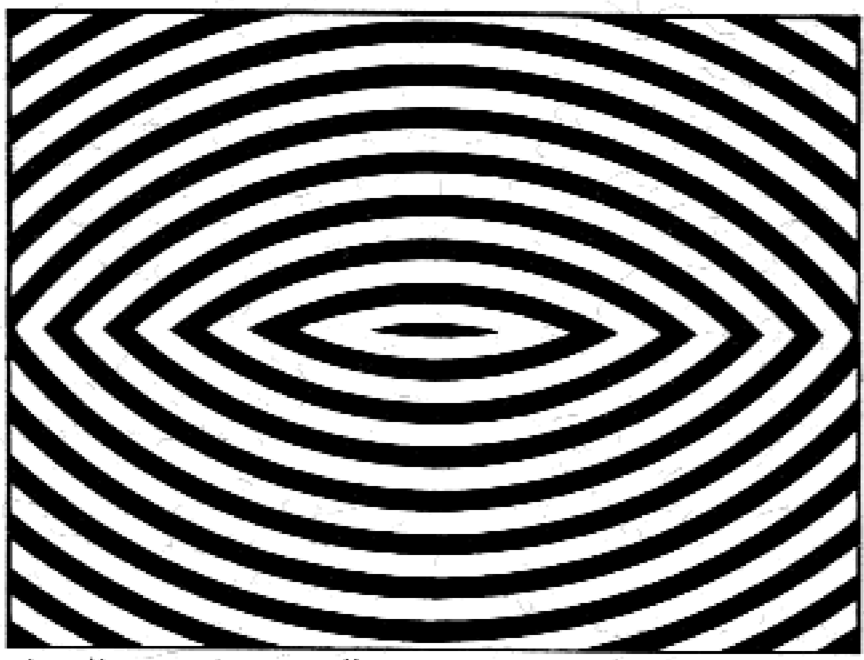
—Unlimited Bandwidth (infra to ultrasonic reproduction)
—Zero Phase Shift (no time alterations)
—Unlimited Dynamic Range (no maximum sound limitation)
—Perfectly Flat Frequency Response (totally neutral tonal balance with no subtractions)
—Absolute Linearity (no distortion due to additions)

As mentioned previously, none of these are completely achievable but they may still be required. To determine which may be absolute requirements, let's examine the human hearing system and what is known about its ability to perceive these characteristics.

The ear has a very rapid rolloff in response below 20 Hz. Output below 30 Hz yields more of a bodily sensation than an actual hearing experience.

The hearing system exhibits a low frequency suppression mechanism such that at lower volume levels, less bass is perceived. As a result of this mechanism, the ear and brain primarily use the information present in the peak bass amplitudes to determine the subjective low-frequency performance.

The ear becomes irritated and fatigued when exposed to sound levels approaching and exceeding 120 dB. At levels above 100 dB, noticeable volume level compression occurs within minutes, and continuous listening at these levels also becomes fatiguing unless pauses occur to reduce these effects.



The human hearing mechanism is only moderately sensitive to bass distortions if they are mainly second and third-order in harmonic content. This is due to their inherent existence in nearly all music and natural occurring sounds. However, the ear is extremely sensitive to non harmonically related distortions and higher-order harmonic distortions, all of which are only present in very specific natural sounds. These then leave their characteristic signature on all reproduced sound.

The ear is also moderately insensitive to gradual phase shifts at low frequencies but is rather sensitive to the amount and rate of change in phase at midrange (voice) frequencies.

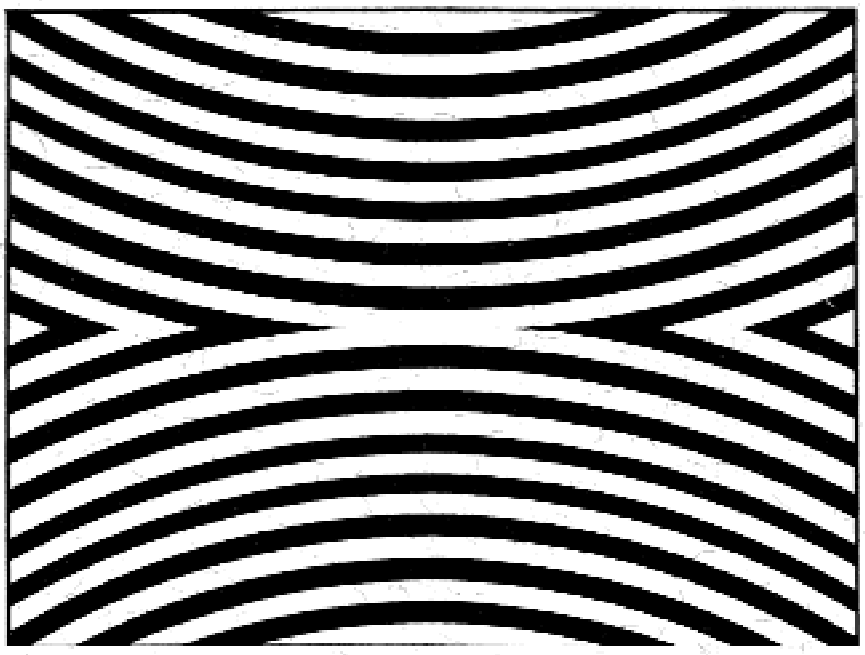
The human hearing system is extremely sensitive in detecting events which are "added to" the original event or occur at times when nothing is present to mask their occurrence. The ear is less sensitive in detecting events which are "removed" or diminished in volume level from the original event.

After reviewing this list of known hearing characteristics, a new list of product performance requirements can be assembled. A product meeting these parameters should then sound "near perfect" because its performance is only imperfect in areas where the human ear is very insensitive. These new performance parameters become:

- Bandwidth down to 10 Hz.
- Reasonable phase shifts are tolerable but should be kept as low as possible.
- The distortion produced must be kept to very low levels. What distortion there is must be confined to low-order harmonics with high-order and non harmonically related distortions virtually eliminated.
- Peak sound levels of over 110 dB are required but not continuously. Levels of 120 dB and higher are infrequently required.
- Performance at high-volume levels is very critical. Therefore dynamic linearity at volume levels above 90 dB is an absolute necessity.
- All amplitude response peaks and sounds, which occur when no sound is called for, must be reduced to extremely low levels.

We have just identified the primary performance requirements of the "near perfect" low-frequency sound reproduction system. But we must remember all of these characteristics are to be met "at the listener's ears", not just in a special environment. There have been some systems capable of meeting these parameters under special conditions. But all fall far short of these goals when evaluated in a typical room at the listener's ears. The reason this subject is not addressed more often is due to one of the recognized purposes of designing and engineering. That purpose is to identify and control the variables in order to ensure a "desirable" outcome when performing a task. Since room effects are seemingly so uncontrollable and varied, the most obvious method of simplification is to leave those factors out of the equation.

Of course, "out of the equation" may also mean "out of the solution" which can keep near-perfect performance beyond reach. Typically, designers use an anechoic chamber to design and test loudspeakers. This chamber is an idealized environment which is actually a room with either a floor but no walls or ceiling (2 pi steradian), or a room which has no boundaries of any type (4 pi steradian). Another approach for testing is to position the speaker 15 to 20 feet off the floor in a very large room (sometimes outdoors) or to use a test setup that only measures the sound coming directly from the speaker. All of these methods share the goal of eliminating undesirable sound by preventing it from being generated or measured "in the test." It seems unfortunate that the listener cannot receive the performance the designer achieved in this special test. It's also unfortunate the room was left out of the equation since it appears to be the real source of the problem. The elimination of the room's boundaries in the testing of loudspeakers greatly improves the measured performance of these products. It also inhibits fidelity improvements at the lowest frequencies because the room interacts noticeably and unpredictably with the speaker, yet is neglected. This does not mean that anechoic testing serves no useful purpose. It is a very valuable controlled environment to perform "before and after" testing of a known or suspected change in product performance. The problem arises when someone assumes that those measured results are representative of how the product performs or sounds in a typical room with floor, walls and ceiling. The industry has mistakenly convinced retailers, consumers, reviewers and even naive engineers that these curves correlate to what is heard and desirable. Other than the average quantity of output available over a frequency range, an anechoic response curve below 100 Hz tells nothing about the quality of low frequency sound reproduction available.



An example of this misconception is a loud speaker product which includes an individual calibrated response curve of its low-frequency response. The response curve correctly states the exact measurement conditions (2 pi steradian) and shows an immaculately flat output of plus or minus one decibel from 30 Hz to 100 Hz. Yes, the curve is very impressive, but it does not represent what the customer will receive when the product is purchased and used.

Since the earliest development of the dynamic transducer, it was known that as the diaphragm moved, it generated two opposite sound waves. It was also discovered that when those two sound waves were prevented from interacting, the low frequency output increased substantially. This effect was later analyzed and found to be a function of the size of the transducer with larger transducers suffering less loss. Since it was only feasible to make and use fairly small transducers, this effect began at about 400 Hz and became progressively more severe at lower frequencies. This was such a dramatic loss of bass that the solution was obvious: use a barrier to isolate the front from the rear wavefront. Unfortunately, it was necessary for the barrier (or baffle) to be very large if the lowest frequencies were to be prevented from being attenuated. The next step in extending the baffle which the transducer assembly could sit (on a table top or floor). This provided an improvement but was still inadequate for the lowest frequencies. The final step was then taken to eliminate the cancellation completely, i.e.. totally close the rear of the baffle and form an "enclosure". Eliminating the cancellation was not the only effect the enclosure had on the sound.

Some of the obvious side effects were noticed even then, with the most notable being an increase in the transducer's resonant frequency. This actually reduced the maximum possible bass output, and only an enormously large enclosure prevented this problem. Likewise, the output at resonance increased which caused a very unpleasant boomy characteristic. After conducting listening tests, it was discovered that the enclosure was also adding sounds of its own to the music. To reduce these effects, designers covered the interior of the cabinet with materials which would absorb sound or prevent enclosure wall flexing. All this was done in an effort to fix a problem caused by a solution to a problem (the enclosure). In some models, designers stuffed sound-absorbent materials into enclosures in an effort to force the box to behave in a more predictable manner. Theories were developed which modeled the performance of sealed and vented enclosure designs, all of which assumed the enclosure was perfect and the measurement room had no walls or ceilings to reflect sound. Meantime, the search for efficiency and greater quantities of sound led the sound reinforcement industry to horn loudspeakers which very successfully achieved those limited but important goals along with directivity control. One very innovative approach, the transmission line, was developed and is still being used today. It was reasoned that the pressure built up inside an enclosed cabinet along with reflected sound off of the interior panels were the major causes of enclosure distortions. The efforts expended to address these concerns by using carefully damped and tapered passageways in the cabinet are commendable. But the complexity, limited dynamic range, and low efficiency of the final products reduced their acceptance. All of these traditional and special designs have one common advantage, they provide a larger quantity of low-frequency output than would be available using the same transducer with no enclosure.

The choice of which design is "better" and by "how much" must first be put into the context that the decision to use an enclosure has already been made. Although the reason for using an enclosure is very convincing, it is the only reason. Several problems associated with enclosures have been mentioned previously. There are six major side effects which cannot be totally corrected when an enclosure is involved in the reproduction of sound:

1.All enclosures shift the fundamental resonance of the transducer upward in frequency, such that less low frequency output becomes available from that transducer. Resonances are characteristics which are generally to be avoided in acoustics, and this upward shift in frequency makes their effect even more noticeable.

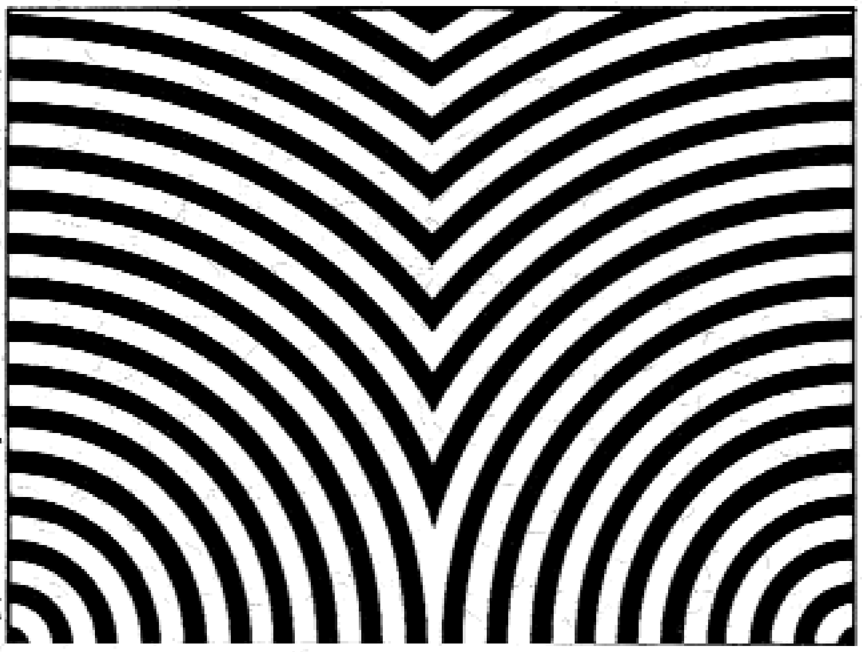
2.Enclosures increase the severity of the transducer's fundamental resonance. This increase in the Q (severity) further accentuates the persistent single-note effect of a resonance.

3.Virtually all enclosures are constructed using flat parallel walls which develop acoustic standing waves between them that are characteristic of the distance between the walls. The longer the distance, the lower the frequency and the more standing waves that occur. These waves cause frequency response and tonal coloration effects which cannot be eliminated, only reduced. They cause the enclosure walls to flex and reradiate their sound into the room, and also reflect sound back upon the transducer's diaphragm causing it to distort the "original sound".

4.Each individual wall in any enclosure possesses mass and stiffness, which together yield a set of natural resonances. They interplay with the speed of sound in the wall material and with the dimensions of the wall to form what can only be described as a myriad of complex wall flexing.

Add to this the fact that all walls have finite rigidity and many are not as rigid as they could be (cost, weight, construction techniques, etc) and we have to wonder if what we hear when we listen to a loudspeaker is really the transducer or the enclosure walls.

In an effort to "improve" the problems of fundamental system resonance shift and increased Q caused by the enclosure, designers have attempted to cancel the resonance by adding another resonance to the system. This resonance cancelling technique (bass reflex) proved to increase the colorations. Bass reflex theory relies on the transducer and enclosure to work as pure and perfect elements in order to reradiate the transducer's rear wave through a port or passive diaphragm (also modeled as pure and perfect elements). Even if all these elements were perfect, the maximum amount of output available through the small ports used in many systems is ridiculously low (less than 90 dB at 1 meter) . Given how well the enclosure performs its function in sealed systems, it is no wonder that, when asked to withstand the much higher internal pressures of a reflex design, it performs so poorly.

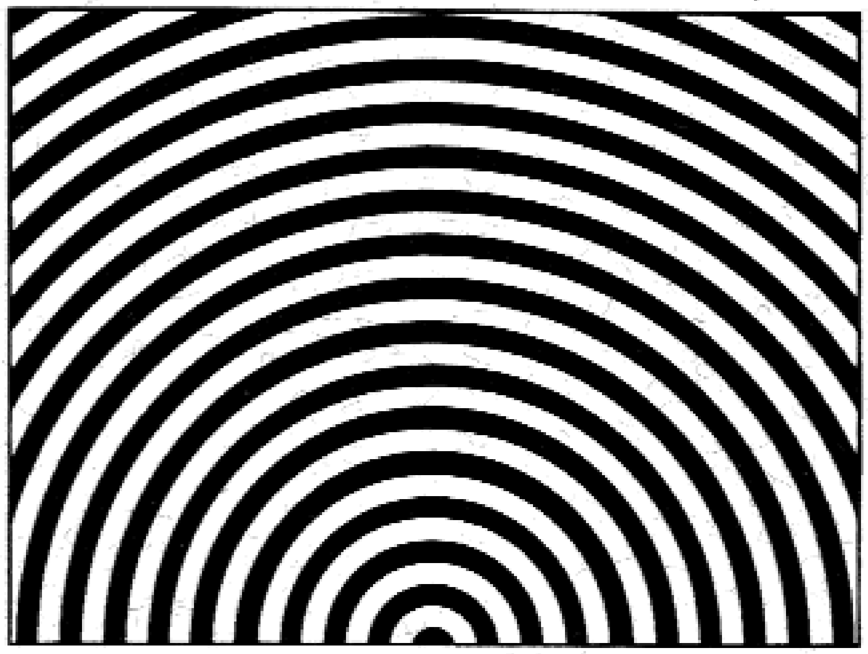


Nearly all enclosed loudspeaker systems are point sources of sound transmission, meaning they radiate their sound uniformly in every direction at low-frequencies. This is most often thought to be a benefit termed as "wide dispersion" or "omni-directional". However, at low frequencies, the room in which the loudspeaker is located has serious difficulties with standing-waves which form between the three sets of parallel room boundaries. In a room such as this, selecting a speaker that generates as much energy as possible, in each of these three directions, would not be the optimum choice. In fact, the point source loudspeaker is the worst possible sound transmitter in this environment, if minimizing the room's effect on the sound is the goal.

As an example of the pronounced effect enclosure wall flexing can have, a typical enclosed 12 inch woofer may have a transducer radiating area of 72 square inches and a cabinet wall area of 2000 square inches This means only 3% of the total external surface area of the loudspeaker system is "desired" or "driven" area. The remainder is undesired potential radiating area that must be rigid to prevent distortion. Just how rigid does it have to be? To keep bass distortion to less than 3% (a seemingly feasible objective) enclosure wall flexing must be kept to less than 0.09% (3% of 3% of the transducer's excursion). Using 90 dB at 75 Hz as an example, we find that the enclosure wall flexing must be kept to less than 0.000043 inch excursion to assure less than 3% distortion. Even when this number is derated (because some walls flex more than others and the central portion of the wall flexes more than its perimeter), the allowable flexing is unbelievably small. Neurological studies have shown that the minimum threshold for tactile sensation of the human hand is also about 40 microinches. This means any cabinet wall flexing on a typical enclosed loudspeaker that can be felt with the fingers is quite likely causing in excess of 3% distortion in the sound you are hearing. This basic behavior of the enclosure is embodied in all enclosed loudspeakers including the very highest performance "state-of-the-art" products. This logically means that very few people have been exposed to low-frequency sound reproduction that is not "clouded" by this effect or by the other five previously listed enclosure effects. Could these effects be the reason many listeners find the sound of full-range electrostatic and enclosure-less dipoles to be more pure and accurate? We believe this is exactly the difference they hear, and may well be the only difference they hear at lower frequencies.

Over the years, designers developed low-frequency sound reproducers which were completely separate from the other loudspeaker components in a stereo system. One of the main reasons for considering the addition of this "subwoofer" product to a system was to increase the overall fidelity and accuracy of the complete system. Yet, when these products are examined for their performance in a typical listening environment, they provide a clear advantage only in the amount of sound they can deliver at low frequencies. Further examination shows that most subwoofers are really just larger versions of the original smaller units they are meant to replace.

The underlying problems in the smaller systems still exist in the larger unit and some of the problems have grown larger along with the size of the enclosure. In a typical room, most mirror the sound of the room and simply "play the room" louder than the original speaker had done. After speaking with listeners who are primarily concerned with musical accuracy and clarity, you find that the addition of a subwoofer to their music system is a very mixed bag.



Usually they find that with music such as pipe organ or large orchestral pieces, the sound is quite "impressive". However, with music not normally considered to have much bass or music which is very transient and percussive, the comments are usually not favorable. These listeners desire to extend accurate reproduction down lower in frequency, down into the sensory "feeling" region. The disappointment these listeners feel is a result of our industry's lack of perceiving one problem (the room) and lack of insight in correcting another of our own generation (the enclosure). But all is not lost. . read on!

When considering what type of transducer to use in creating the most accurate low-frequency sound reproduction system available, there are only two realistic alternatives: electrostatic or electrodynamic.

The electrostatic has a reputation for accuracy and uncompromising performance. Yet for those who own them (we have) and for those who design them (we have), the reputation may be somewhat overstated at the frequency extremes, especially the lower extremes. There are few equals to the electrostatic in the midrange (voice) region. However, at low frequencies the story is different. Certainly, the electrostat's difficulty in developing large diaphragm excursions may be part of the reason. But are there other reasons? This question must be answered if the optimum transducer is to be selected on true merit and not mystique. The following is a list of generally agreed upon characteristics of the electrostatic transducer.

—The maximum force per unit area available to move the diaphragm is extremely low.

—The force required to move the diaphragm is also very low. This is because the mass per unit area can be made so low that at low frequencies it be comes insignificant compared to the air's mass.

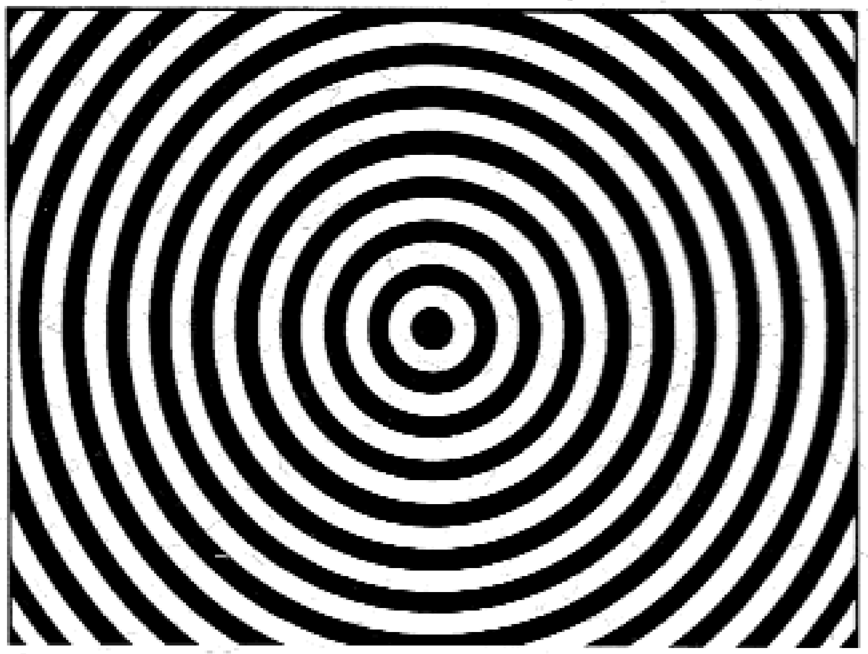
—Large air volume displacements require a very large radiating area since available diaphragm displacement is very limited.

—Generally, electrostats display rather abrupt and pronounced misbehavior when driven beyond their designed capabilities. Since diaphragm excursions are limited for electrostatics, and because excursion requirements increase dramatically at lower frequencies, reasonable low end response and dynamic range are extremely difficult characteristics to achieve.

—The extremely low mass of the diaphragm, its lack of rigidity and the low force per unit area combine in preventing the effective use of a rear enclosure in all but very special cases. This also means the transducer is very sensitive to changes in the air load it "feels".

—The constant-charge, balanced, push-pull operation of an electrostatic offers extremely low distortion possibilities.

When a large electrostatic diaphragm is used to reproduce low frequencies in an environment where extreme variations in the sound pressure exist (due to room modes) , the transducers diaphragm will not have the ability to overcome these changes in its airload. It will passively respond to the room rather than the input signal. This is termed acoustic output impedance and the electrostatic has an extremely high output impedance. Also, the room's pressure may not be uniform on the diaphragm and since the diaphragm has virtually no rigidity, it again succumbs and flexes. Under large excursion conditions, the edges of the electrostatic diaphragm experience stretching because no separate edge suspension exists. Eventually this may lead to fatigue and either performance changes or failure.



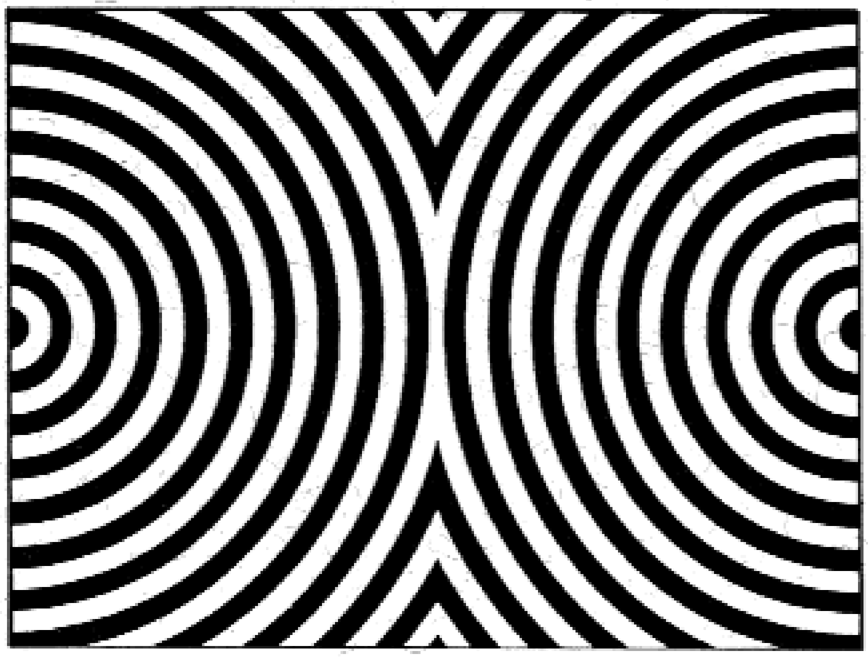
The key to understanding the optimum usage for the two types of drivers (ESLs and moving coil) is to realize that the requirements for higher frequency reproduction coincide with the strengths of the electrostatic driver. In turn, the requirements for lower frequency reproduction match well with the strong points of the moving coil driver. We believe some of the most accurate loudspeakers ever made and still being made are electrostatic systems. No disrespect is intended in relaying this information on extreme low-frequency behavior. The historic performance of high-quality electrostatic systems demonstrates that they have always been on the leading edge of technical development and high performance. Owners of these superb systems will find their concern for accuracy well matched to the capabilities of the device we will soon describe. As pointed out earlier, one advantage to open back loudspeakers is their elimination of the enclosure problems by simply eliminating the enclosure.

There is yet another advantage when the cabinet is removed. The resultant dipole system no longer radiates sound as a point source. The system now radiates its sound in what is called a cosine pattern. The sound source becomes two sound sources located in the same place.

One source radiates a sound wave frontward while the other radiates a "reverse polarity" sound wave rearward. An interesting and useful phenomenon occurs in the plane of the transducer (top, bottom and on both sides). The sound from the frontal wave is exactly equal to the sound from the rear wave. Since they are reversed in polarity, they completely cancel each other. This might first be thought of as a loss of precious bass output. But, this effect only occurs in directions where we are not listening. The frontal wave we hear remains unaffected. This cancellation in the sound output in four directions yields a tremendous improvement in tonal accuracy and response smoothness when analyzed in a typical listening room. If the decision to develop a subwoofer is to be made, two other key concerns must be addressed. The one that stirs the most controversy is: Stereo vs. Mono Bass. This choice can be further divided into stereo dual-source bass, mono dual-source bass and mono single-source bass. The reason for this further breakdown is that, as we will discover shortly, the number of sources located in the room is more critical to performance than mono or stereo operation. The most logical decision would be to preserve "full stereo" reproduction as was the intent of the original program material. That seemed logical enough, but was it? Here is what was found after some investigation:

Virtually all bass below 100 Hz is purposefully or unknowingly recorded monophonically. That which is not, is almost never done purposefully, but is accidental. The reason it is recorded monophonically is that loudspeakers which do not operate "in unison" below 100 Hz produce unpredictable losses in bass output due to room modes, speaker placement and the listener's location in the room.

Sound becomes progressively more difficult to localize as its frequency is lowered. This means human hearing is unable to distinguish where a sound is coming from at extremely low frequencies. Sounds above 200 Hz tend to be those used by humans to locate the source of an event. Frequencies below 100 Hz tend to give us a sense of size, power and space but not direction.



When monophonically recorded bass program material is replayed through two physically separated loudspeakers, whether driven monophonically or stereophonically , a listener must be physically positioned an equal distance from each speaker or the sound heard will not have the intended tonal balance, bass power or impact.

Even when the listener is listening equi-distant from the two loudspeakers, the bass performance will still be compromised unless the room is exactly symmetrical and the speakers are placed in perfect symmetry in the room. The reason is that even small amplitude differences caused by the room's resonant modes will shift the phase of one speaker's output quite differently than it will the other speaker's output unless everything is perfectly symmetrical. This again leads to a loss of impact and bass power which were present in the original recording.

The luxury of having to place only one low-frequency loudspeaker in a listening environment is more than just a size, logistics, aesthetic and economic benefit. It also means only one thoughtful decision is required for optimum sound reproduction rather than two, thus eliminating interaction and needless side effects.

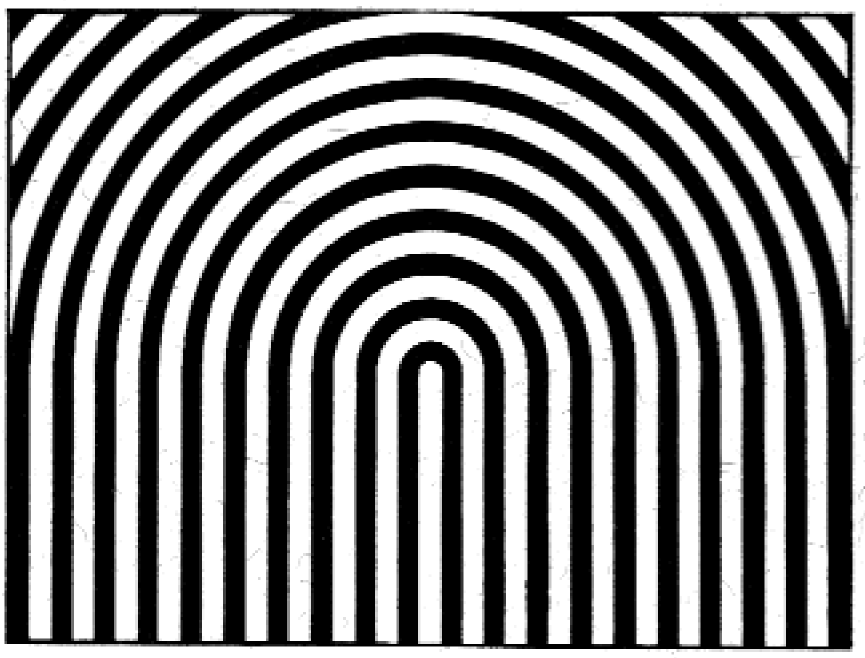
If these factors are taken seriously and careful listening tests are performed to confirm their importance in the sound reproduced, the results are obvious. The "logical" decision to "preserve stereo" will consistently produce inferior sound at low frequencies. Remember, we are not dealing with the concept of a dynamic range improvement brought about by using two speakers, because the device we are contemplating will need no additional dynamic range. Remember the single-source low frequency decision is not a compromise. It is the optimum decision with dual-source being the performance compromised approach.

The basic groundwork has now been laid. The compromises in enclosed bass loudspeakers force us to pursue an encloseureless approach if the ultimate bass reproduction system is desired. Though electrostatic transducers offer superb acoustic performance potential, they are not ideal, nor even preferable, for extremely low-frequency reproduction unless major changes to their design are implemented. The system must be a dipole source to reduce the performance compromises of a closed box source design and to maximize the demonstrable impact and power of the reproduced sound. Lastly, the electronics should be kept to a minimum which meet all the requirements for low-frequency accuracy. These decisions were consolidated and formed a target design criteria for a device which would embody the highest performance attainable at low-frequencies. The initial prototype was then developed, and testing proceeded to evaluate its performance. The results from measurements and listening tests were astounding. All those involved indicated the product was superior to anything available. ✳

ANTI-NOISE

The concept of controlling noise by generating a mirror-image, out-of-phase signal is hardly new; it has been in commercial and industrial use for a number of years. Indeed, the patents covering this technique date back more than 50 years. What is new is the application of antinoise principles to consumer products. The day is approaching when even the annoying whine of a blow-dryer or the cacophony of a lawn mower—annoyances of late sleepers everywhere can be nullified.

Sound travels through the air in a classic pressure waveform. Its pitch is dependent on frequency, or the number of waves that reach a given point every second. Between the pressure peaks is a low point—the trough of the sound wave—and when a peak and trough of equal frequency and intensity (amplitude) intersect, it results in almost total noise cancellation. As a natural phenomenon, it has long been known to concert hall designers. This cancellation zone of silence—where reflected sounds in a room meet with a generated signal of equal frequency but 180 degrees out of phase—is known as a node point.



Headphones, as used in the systems for MRI scanners and pilots, prove to be an ideal method for deploying antinoise, because the enclosed sound environment can be tightly controlled. In the Bose aviation headset, a dramatic reduction in noise is accomplished by insulating the headset from outside sources and doing both the monitoring and cancelling inside the headset itself.

With increasing regularity, auto-show concept cars include active noise cancellation as one of their advanced features.

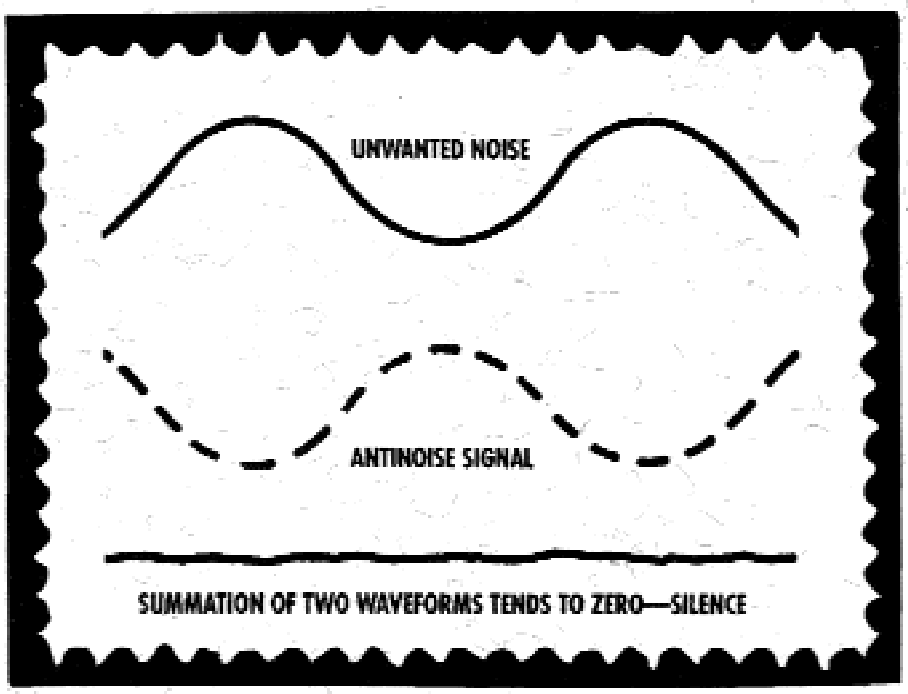
More than any other consumer product, the automobile is rich with possibilities for antinoise applications. Recent projects undertaken by Lotus Cars in England are among the first to catch the public's attention. Though passenger comfort may seem incentive enough to spur antinoise research for cars, the primary motives are somewhat less altruistic. By eliminating exhaust-flow restrictions created by the multiple baffles and labyrinths in a conventional passive muffler system, improved fuel economy can be achieved.

In some ways, this is the ideal application for antinoise: The noise is contained in a duct (the exhaust plumbing), it is repetitive and predictable, and a great deal of its spectrum falls within the low-frequency range where active noise cancellation is most effective.

It is a fortunate quirk of nature that antinoise works best on the low end of the scale where other methods aren't nearly as effective. Traditional passive sound-deadening techniques—thick insulation, absorbent padding—efficiently quell high frequency signals, easily dissipating their energy. A low-frequency rumble is much tougher to deal with, though. Fortunately, the lengthy low-frequency wave offers an antinoise signal a wide working range for cancellation.

For example, the length of a sound wave is determined by dividing its speed by its frequency. Roughly speaking, a sound with a frequency of 100 hertz traveling at sound's typical 1,200-feet-per-second pace in sea-level air has a 12-foot-long waveform.

Total cancellation occurs where the peak and trough intersect perfectly—where the sound waves are exactly 180 degrees out of phase; some degree of cancelling will be achieved, however, even where the waves are not perfect mirror images. On a short, high-frequency wave, the zone of cancellation is a very small area in space, so small that a 10,000-hertz whistle may be nullified at your left ear but not at your right. Low-frequency sounds, however, offer a peak that presents a much larger target for cancellation. Walker Manufacturing Co. of Racine, Wis., a major supplier of conventional auto mufflers, has steadily developed the technology in recent years under a joint venture agreement with NCT, concentrating on frequencies less than 500 hertz.



Introducing the antinoise signal also presents special engineering challenges. An engine's exhaust system with high temperatures and corrosive water vapors, presents a rather unfriendly atmosphere for antinoise components. In the air-conditioning duct of an office building or the exhaust stack of a factory, it's simple enough to mount the loudspeakers in a T-shaped junction and inject the antinoise signal directly into the pipework. In the systems under development at Walker, however, the antinoise signal is carried in a sealed pipe that surrounds the actual exhaust system. There is no contact between the hot engine gases and the twin four-inch antinoise speakers. The opposing signals for the exhaust and the annular pipes meet and cancel only after reaching their rearmost outlets.

The basic requirements of an antinoise system are disarmingly straightforward. The noise source is monitored to determine its pitch and loudness, and this data is fed to a processor for analysis so the mirror-image signal can be created.

A downstream microphone listens to the cancellation results to provide a correction factor.

The process of synthesizing the correct signal may be achieved in many ways. One obvious approach is to simply place a microphone at the noise source to collect the frequency and amplitude data, feed this signal to a high speed processor, and generate the correcting signal. Early experiments with this technique faltered because the microphone was unable to distinguish the source noise from the antinoise signal, hopelessly locking into a feedback loop.

Highly directional microphones and speakers were tried, but the real breakthrough came in Britain from professor George Chaplin. He replaced the microphone with a tracking signal sensor that supplies a synchronizing input—a tachometer pickup, for example—allowing the processor to predict sound.

A microphone at the outlet evaluates the residual noise and provides a correction signal to the processor that generates the antinoise. More recent advances in electronics enable the noise to be directly read at the source without confusion from the antinoise signal.

A device known as the Infinite Impulse Response Adaptive Filter, developed by Digisonix, a division of Nelson Industries in Middleton, Wis., allows noise to be sampled at its source, or upstream, rather than at the downstream (outlet) position. A down stream microphone provides the processor with an error-correction signal based on the final noise output.

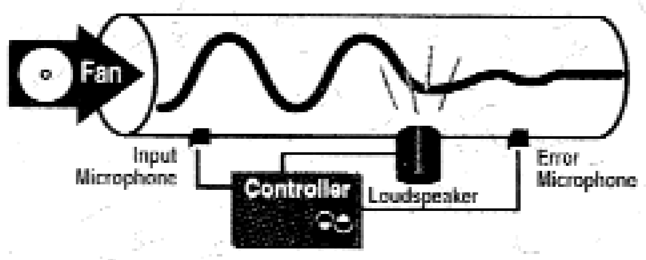
The advantage is in response and the ability to cope with random noise, according to Nelson's vice president of research, Larry Eriksson. "With our system, the fact that a sound is not repetitive does not preclude cancellation. We can do broadband noise as well as we can do a continuous tone."

Beyond the obvious uses—exhaust silencers, air-conditioning fans, and the like—there's a multitude of rich possibilities in the more subtle sources of disturbance. By applying noise cancellation techniques to engine supports, vibration can be reduced before it ever reaches the passenger compartment. The active engine mount is a goal being pursued by virtually every car maker today, a new-age "electronic" balance shaft that extracts no engine power for drive and works over a wide range of engine speeds. Active mount technology crosses into an area where disturbing signals are not exactly noise, but rather are low-frequency vibrations that intrude on comfort and hasten fatigue over the course of a long drive.

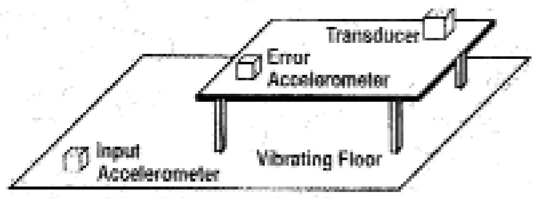
Simply, the body shell of a car is a resonant cavity—comparable to the hollow body of an acoustic guitar—where a perceptible "booming" is amplified. Disturbance sources include the natural vibrations created by unresolved shaking forces inside a four-cylinder engine or the harmonics generated by a rough road surface.

In what must be regarded as a classic paper in the context of the active control of sound, Olson and May (1953) introduced the "electronic sound absorber". Although the title of the paper is something of a misnomer, Olson and May foresaw many of the applications of active control techniques that are now coming close to practical use. The device described in the paper is illustrated here. The "perspective view" of the apparatus illustrates well the state of electronic technology at the time the paper was written.

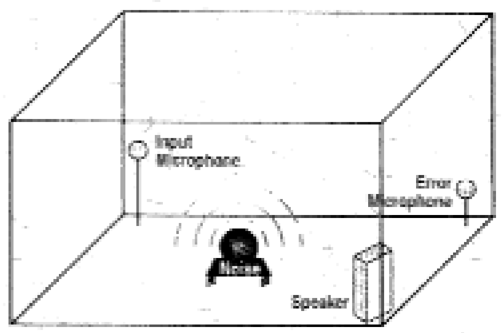
Ducted Noise from Fan or Pump



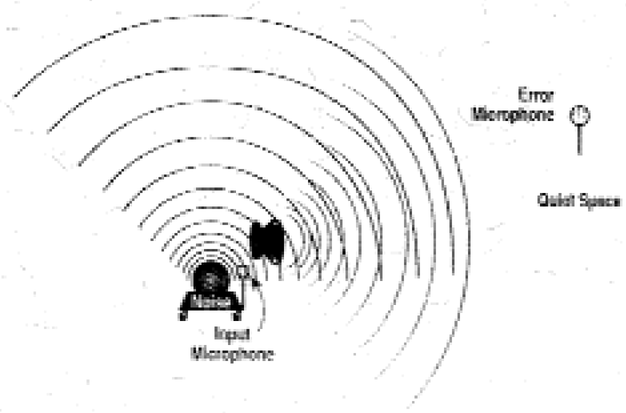
Structural Vibration Control



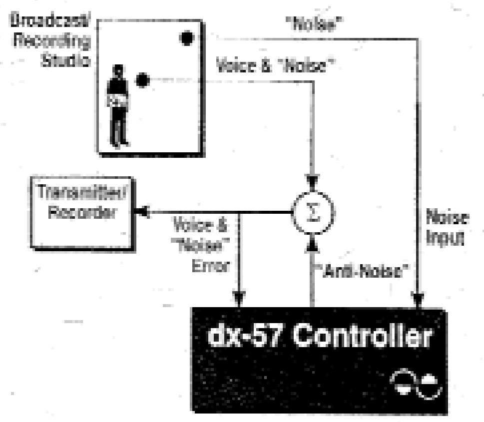
Enclosure Silencing



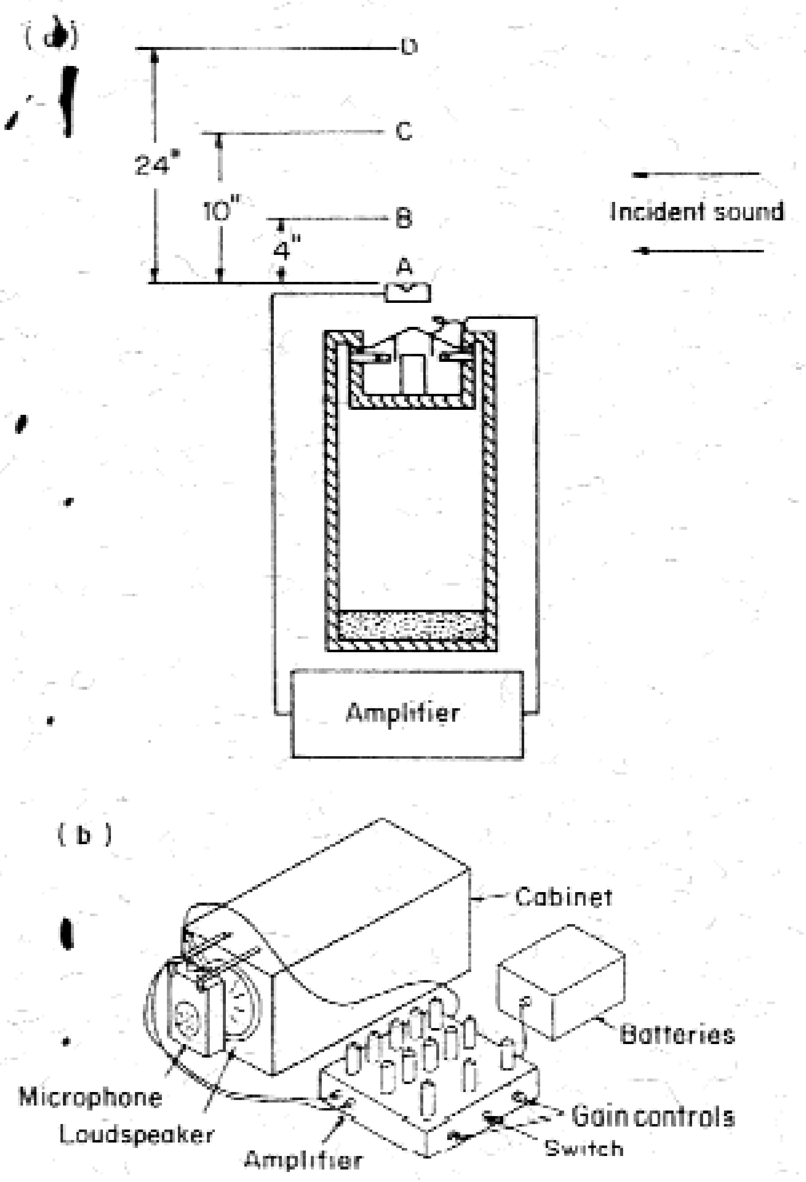
Open Field Cancellation



Communications System

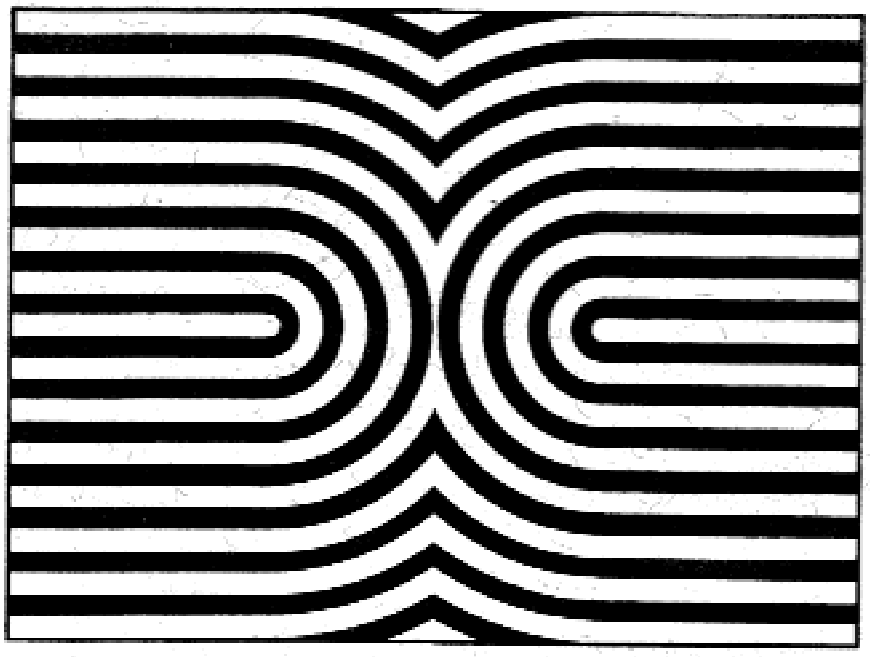


An "electronic microphone" (in which "the impinging sound vibrations directly control the electron stream in a vacuum tube") was used to derive a signal proportional to sound pressure. The microphone had a flat frequency response down to 0 Hz and a phase response that was "less than two degrees from 20 Hz to 400 Hz". The signal was amplified by using a battery powered valve amplifier. The amplifier output was used to drive a loudspeaker with a "high impedance voice coil" which avoided the necessity of a transformer between the vacuum tubes of the amplifier and the loudspeaker. A transformer would introduce a "considerable phase shift" in the low frequency range and this was to be eliminated. In addition, the response of the amplifier was made to roll off at frequencies above about 400 Hz to avoid "positive feedback". The overall operation of the device was made to ensure that the phase shift could be minimized between the input to the microphone and the output from the loudspeaker. Under these circumstances a simple inversion of the microphone signal could be introduced such that the loudspeaker output was 180° out of phase with the microphone input. If a high gain of the amplifier was used, the device would then act as a "sound pressure reducer"; the detected sound pressure was fed back with a phase inversion and high gain in order to drive the pressure fluctuation towards zero. Driving the sound pressure in the vicinity of the loudspeaker to zero of course ensures that the loudspeaker neither radiates nor absorbs any energy under steady state conditions. This was recognized by Olson and May but they suggested that the device could also absorb sound by "designing the system so the proper phase relations are obtained". They also saw that this was difficult to achieve in practice and suggested that if the region of the loudspeaker and microphone was surrounded by an acoustically resistive screen, then this could be made to dissipate energy. Olson and May suggested that in general "the problem in low frequency sound absorption is to provide an acoustical impedance of relatively small value so that the volume current which introduces the sound absorption will not be limited by high acoustical impedance". Thus, Olson and May reasoned that the feedback system producing low sound pressure could provide a low impedance for terminating the dissipative acoustic impedance provided by the surrounding screen. However, most of the applications suggested by Olson and May involved the use of the device as a "spot sound reducer".



Since the earliest development of the dynamic transducer, it was known that as the diaphragm moved, it generated two opposite sound waves. It was also discovered that when those two sound waves were prevented from interacting, the low frequency output increased substantially. This effect was later analyzed and found to be a function of the size of the transducer with larger transducers suffering less loss. Since it was only feasible to make and use fairly small transducers, this effect began at about 400 Hz and became progressively more severe at lower frequencies.

This was such a dramatic loss of bass that the solution was obvious: use a barrier to isolate the front from the rear wavefront. Unfortunately, it was necessary for the barrier (or baffle) to be very large if the lowest frequencies were to be prevented from being attenuated. The next step in extending the baffle was to fold it backward so it would take up less physical space and provide a platform upon which the transducer assembly could sit (on a table top or floor).



This provided an improvement but was still inadequate for the lowest frequencies. The final step was then taken to eliminate the cancellation completely, i.e. totally close the rear of the baffle and form an "enclosure". As is well known, dipole radiators with a finite baffle size lose radiation output below a critical frequency due to front-to-back cancellation of sound energy.

This critical frequency is determined by the average baffle dimension. Cancellation (at the rate of 6dB/octave) is effective for frequencies whose half-wavelength is larger than the average baffle dimension. At 50Hz, the half-wavelength is 11', so that dipole radiators must be large in size to maintain decent bass efficiency.

Now that we have both a historical and mildly technical perspective of these principles concerning the active control of sound, lets discuss how they will apply to ESLs. In particular, how can we increase ESL bass using these fascinating concepts.

We all realize that enclosing the rear wave of an ESL is very impractical since it not only raises the "Q" but, more importantly, the ESL loses tremendous volume. The enclosed air must now be compressed by the weak physical diaphragm power of the ESL. But if we consider the backwave to be "undesirable noise" and apply active noise control principles to reduce the cancellation effect, we should increase the quantity of desirable frontwave ESL bass.

And since the backwave is acoustically predictable, that is, it is exactly the same as the frontwave only 180° out of phase, we can eliminate the microphone and other complex equipment associated with noise control approaches. So we need not deal with unpredictable noise sources.

All we need to do is feed a duplicate signal (music) from the ESL amplifier to a second radiating source that is 180° out of phase with the rearwave of the ESL.

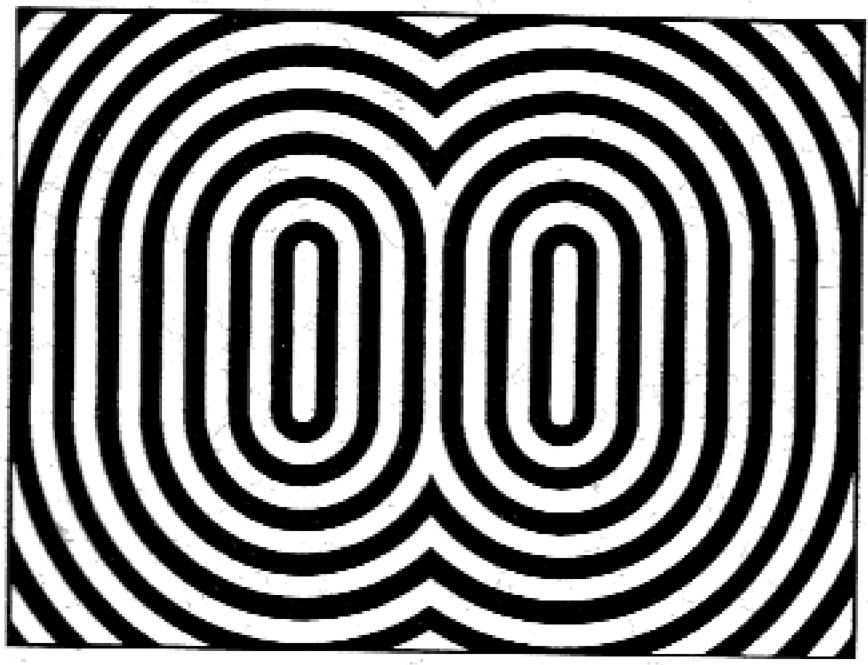
But this is not as easy as it may sound since the second source must be of sufficient power and size plus all phase corrections must be implemented to avoid distortion of the positive or frontal wave.

The solution is still a difficult one to realize unless we employ several other concepts. The next three chapters will cover each aspect of the solution in a more specific manner. So read on!

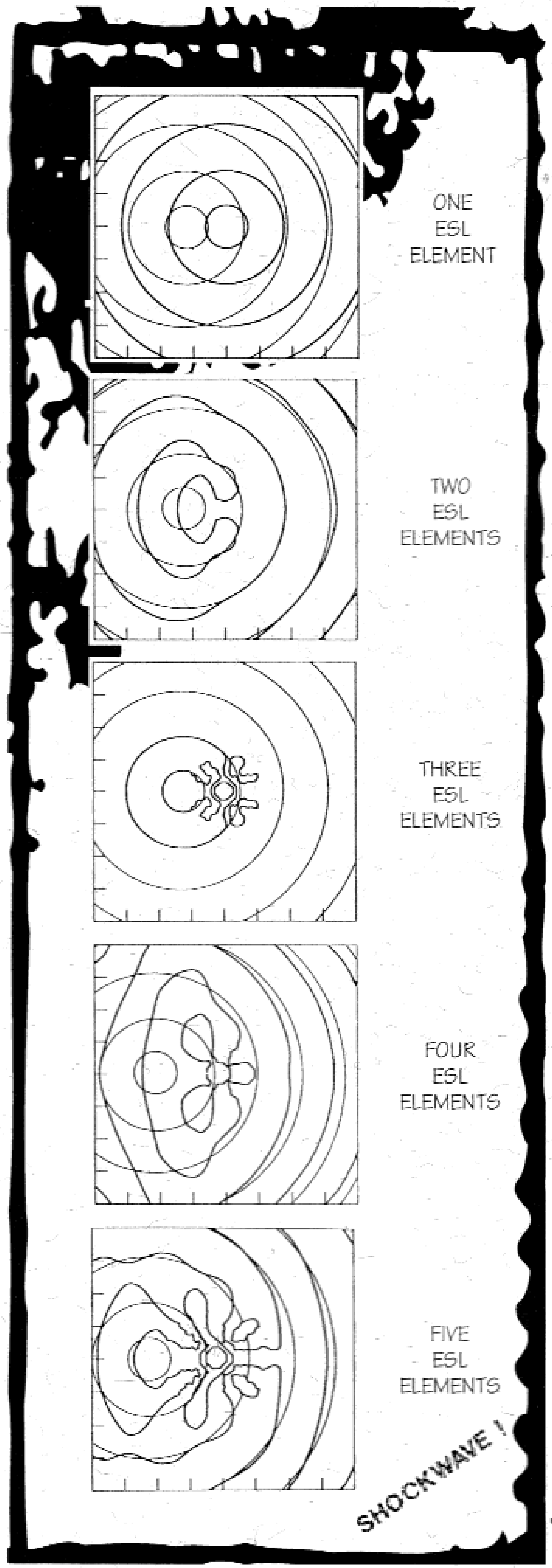
SOLUTION 1

REARWAVE DESTRUCTION

In this section we will begin to address the problem of the active control of free field acoustic radiation. Our starting point is an investigation of the nature of the sound field produced when a simple "secondary" point monopole source is introduced in order to control the radiation of an existing "primary" point monopole source, both sources radiating sound at the same single frequency. The form of the resulting sound field has long been understood and in the particular case of the two sources being of equal and opposite strength and separated by a distance that is small compared to the wavelength of the sound radiated, the sound field produced reduces to the well known dipole field. It is well known that to produce substantial reductions of the sound field radiated by a simple source, we have only to introduce in close proximity a simple source of equal and opposite strength. However, a closer examination of the problem reveals that this strategy is not necessarily the best approach, and that to minimize the total sound power radiated by the source combination a subtle variation of the secondary source strength is necessary. Many of the same physical processes observed in the one-dimensional case are also evident in the equivalent three-dimensional problems. For example, we shall again see the capacity of a secondary source both to absorb radiation from a primary source and also to prevent radiation escaping from a primary source. The latter mechanism of "loading" occurs through the secondary source producing the appropriate pressure on the primary source at the appropriate time. Since there is a finite time taken for sound to propagate from the secondary source to the primary source, the secondary source has to anticipate the output of the primary source.



Analysis conducted in the frequency domain often reveal such "non-causal" action of the secondary source when the solutions are transformed into the time domain. The case examined is of a point primary source whose source strength fluctuation is a stationary random process. A secondary source is introduced in order to control the field. At field points closer to the secondary source than the primary source, the sound pressure fluctuation can be cancelled perfectly through a delayed action of the secondary source. For field points further from the primary source than the secondary source, the secondary source must produce an optimal prediction of the primary sound pressure fluctuation in order to minimize the time-averaged value of the pressure fluctuations. It is demonstrated that for stationary random fluctuations in the primary field, the best achievable performance of an active controller will often depend on the statistical properties (and thus predictability) of the primary pressure fluctuation. This simple result shows that if a single secondary source is used to drive the pressure to zero in the far field of a single point primary source then, provided the two sources are separated by less than one-twelfth of a wavelength, attenuation of the entire far field will be produced.



We will see that ensuring that primary/secondary source separation distances are much less than the wavelength of the sound being radiated is a consistent requirement for the successful global control of sound fields. Before leaving this discussion of the simplest of all three-dimensional active control problems, it is interesting to observe the form of the resulting sound field by the addition of parallel ESLs relative to the acoustic wavelength and various chosen points of cancellation in the far field (see illustration at left). Five parallel elements prove to be optimum.

In determining the global effectiveness of active control, the total power output of given source arrangements is a very useful measure. The power output of an acoustic source is given by the integral of the acoustic intensity over a surface surrounding the source. In the far field, the mean squared pressure is a measure of the acoustic intensity. Thus, the acoustic power is effectively an addition of all the contributions to the mean squared pressure at all locations in the far field. In order to calculate the total power radiated by an array of sources it is usually necessary to perform an integral of the mean squared pressure over this far field surface. This integral is usually fairly easy to evaluate in relatively simple cases, but becomes more difficult as the complexity of the source arrangement increases.

The primary source power output can be modified by the addition of the secondary source pressure field which changes its ability to radiate power. If the appropriate pressure is to be produced on the primary source then the secondary source must clearly emit its signals at a time prior to those emitted by the primary source. Similarly, the second possible mechanism of power output reduction is by sound power absorption. If the output from a primary source is to be absorbed by the secondary source then the secondary source must clearly act at a time after the emission of the relevant signal from the primary source.

This aspect of the production of optimal results is of little consequence when one is dealing with purely periodic signals. In our consideration of the simple combination of point primary and secondary monopole sources. We have shown that there is a well-defined global optimum for the minimization of the total source power output.

We have already noted the particular form of far field sound pressure produced by the interference of two sources of equal and opposite strength when these are spaced a small distance apart relative to the acoustic wavelength at the frequency of interest. We have seen that the dipole source has a particular significance in the active control of one-dimensional sound fields, where in combination with a monopole source it can act to produce unidirectional radiation.

This is also the case when a continuous layer of monopoles and dipoles is used to control three-dimensional radiation. As a prerequisite to this discussion it was useful to describe the nature of the point dipole field. The point dipole results when the product of the strength of the component sources and their separation distance is held constant as the separation distance is allowed to vanish. A more extensive study Of this approach has been undertaken by Thornton (1988) who examined a number of primary/secondary source geometries and consistently found that close to optimal results could often be achieved with a relatively small number of elements. Thornton found for the case considered here that negligible further reductions in power output were produced by increasing the number of elements from five to twenty.

When directly applying these principles to ESL dipoles, it is apparent that multiple elements are necessary to produce significant out-of-phase power that will destroy most of the ESLs backwave that is 180 degrees out-of-phase with the frontwave

Our experiments show that one primary and four secondary parallel ESL elements are optimum for rearwave destruction that approaches 75% efficiency.

POINT/LINE PERPLEXITY

Many enthusiasts are confused, if not dismayed, over sensitivity measurements of ESLs they have constructed. Lets explore this topic, briefly, in the hope of gaining some insights.

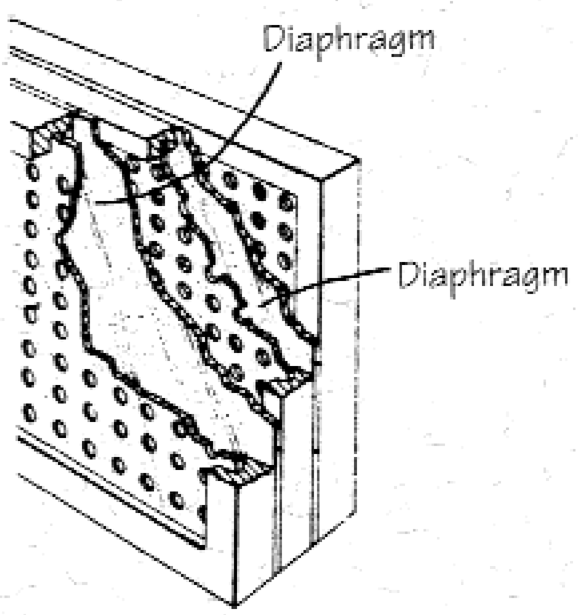
All ESLs are considered vertical acoustic line sources and all sound energy radiates from a hypothetical vertical line. The length of this line is considered large compared to the wavelengths it produces. The major characteristic of a line source is that it disperses energy only in the horizontal direction. The volume over which the sound is dispersed looks like a cylinder—it has a flat top and bottom. Most conventional or "box" systems emanate energy from a hypothetical point. The point is much smaller than the wavelengths it reproduces. The point source disperses energy equally in all directions. Now comes the key to understanding the differences between these two sources. Sound level varies as the inverse of the distance to the line source whereas, it varies as the inverse of the square of the distance to the Point source. In simple terms, the sound intensity drops off significantly slower in a line source than than with a point source. So when we measure the sensitivity using the standard test procedure (SPL when one watt is applied to the speaker and measured at a distance of one meter) the results never seem to match when we move back to a normal listening position. And this position, of course, is much greater than one meter. Our ears are not deceived by this test and most will exclaim that the measured sensitivity at one meter does not match the perceived level at normal listening distances. They will say: "It doesn't sound like an 84db speaker back here." In practice, the line source obtains a 3db advantage over the point source with each doubling of distance from the source. Typically, a line source speaker with a measured sensitivity of 84db has the same sensitivity as a 90db point source speaker at 10 to 13 feet depending on room conditions. So the moral of the story is: measurements should be taken at normal listening distances when comparing point sources to line sources. Things sound a lot better now!

SOLUTION-2

COMPOUND DIAPHRAGMS

In general, electrostatic transducers have been limited with respect to power output capability, particularly in the lower range of frequency response, because of narrow spacing between the active elements. With the given spacing, only a given voltage may be applied due to breakdown of the air dielectric, and thus a limit to the force which can be developed is established . Application of this class of transducer has therefore been limited to designs that can accept this limitation. In particular, this limitation is more noticeable at low audio frequencies, since it is desirable to totally enclose the back of the speaker with a reasonably sized enclosure, and this in turn implies that high forces be developed to move the diaphragm against the air loading. The foregoing will be more apparent in considering a system, by way of example, for converting audio frequency electrical energy into acoustical energy. It is axiomatic in ESLs that to realize equal power output over the audio frequency range, a much greater mass of air must be moved at the lower frequencies and it is this reason that low frequency loudspeakers are much larger and more heavily constructed than high frequency loudspeakers. In any case, the low power output limitation is the result of the limited maximum excursions demanded of the flexible diaphragm.

The diagram shows a plurality of flexible diaphragms mounted in a system of stacked frames with rigid plates interleaved between. Both the diaphragms and plates are of conductive material and plates are provided by using perforated metal.



With such an arrangement alternate ends of a center tapped winding of a transformer are respectively connected to successive plates. Suitable bias is supplied to the diaphragms by connecting successive diaphragms to a static high voltage source. By applying an alternating input signal to the transformer, the diaphragms are each subjected to electrostatic forces in the same direction with the direction being reversed for all diaphragms for successive alteration of the signal. The result is that the force contribution of each diaphragm is additive to that of the others. The increase in force is provided by the laminar type of construction without differences in displacement from diaphragm to diaphragm. Except at very high frequencies where provision must be made for wave propagation velocity, typical increases of force are on the order of 6db with each doubling of diaphragm elements.

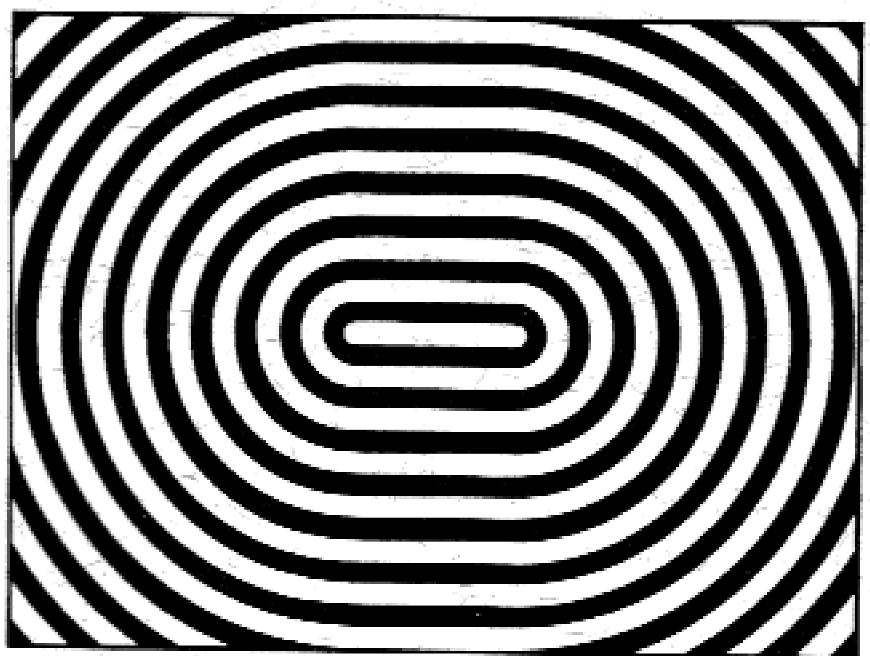
Parallel diaphragms go back at least 50 years, although commercial implementation has been sparse due to the following reasons: 1. Costly and complex construction. 2. Phase problems in the upper frequency range. 3. Problematic increases in the diaphragms resonant frequency with associated increases in the "Q".

SHOCKWAVE uses multiple diaphragms, but in a simpler manner, as 5 independent ESLs are placed in parallel without the need for the middle perforated plate. This also allows for greater spacing of the elements so that rearwave destruction may be achieved.

SOLUTION 3

RESONANCES & RESISTIVITY

An extremely important design consideration for the construction of SHOCKWAVE is the amount of tension on the diaphragm. In addition to such factors as resonant frequency and output magnitude, diaphragm tension also affects transducer sensitivity in at least two ways. Within limits, less diaphragm tension provides greater reception sensitivity. Also, incorrect diaphragm tension may introduce stress patterns into the diaphragm causing said diaphragm to wrinkle which will affect the ability of the diaphragm to uniformly vibrate. Such nonuniform diaphragm movement will directly affect transducer efficiency and therefore indirectly affect transducer sensitivity. Let us examine some aspects of the mounting of the diaphragm. The diaphragm must be tensioned to give a positive stiffness sufficient to counteract the negative stiffness or force resulting from the constant charge. The acoustical loading on either side of the diaphragm is low at low frequencies. With the thin, light plastic materials presently available for use, the elastic restoring force may not be sufficiently stable with respect to time to prevent eventual collapse. Of course, greater stability can be obtained by sub-dividing the diaphragm into smaller areas bounded by supports, but this solution results in higher resonant frequencies. The directivity pattern can also be altered by mechanical and acoustical treatment. But any dampening of the rearwave results in lower output and sensitivity.

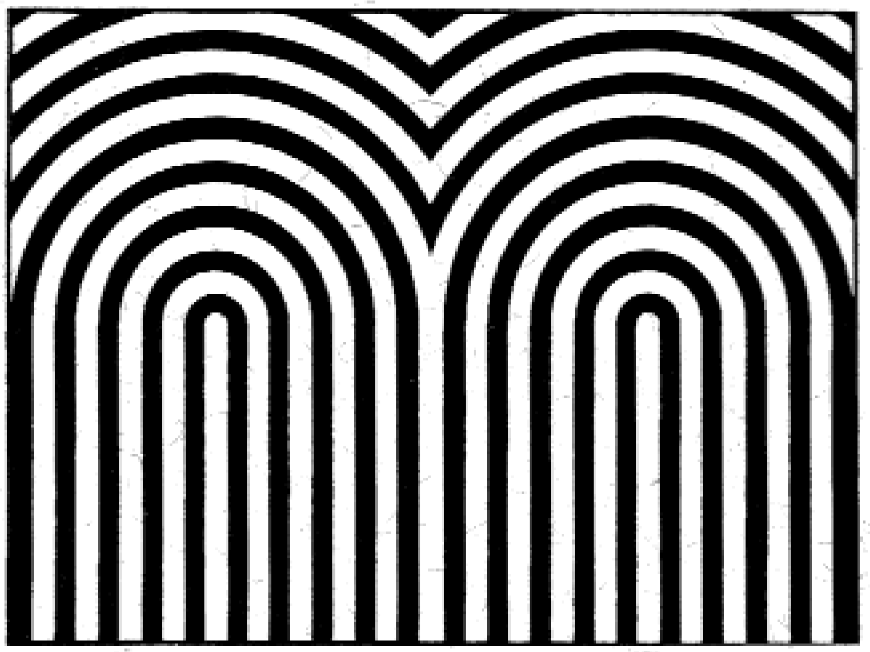


We know that the mechanical Q of the transducer plays an important part in controlling the frequency response of the radiated power. The calculated value of the mechanical Q is generally lowered owing to the mounting loss of the transducer. Thus mounting is very important, since this additional dampening reduces the intensity of the sound radiation. Let us now study the vibration of a stressed membrane which will be applicable to the motion of the SHOCKWAVE diaphragm. One of the classic examples is a kettledrum, which we will consider briefly as it is applicable to our loudspeaker design. Striking one of the stretched membranes causes alternating compression and expansion of the enclosed air, which exerts force on both membranes and modifies their modes of vibration. When the velocity of the transverse waves in the membranes is low compared with the velocity of sound in air, then the effect of the motion of one part of the membrane transmits very rapidly through the air to affect the other parts, and so the reaction of the air is uniform over the membrane's surface. When the membrane vibrates, due to the adiabatic pressure-volume variations then excess pressure is undesirable since this pressure always opposes the displacement of the diaphragm. The load offered to the diaphragm is expressed as a resistive term (additional reactive load is added when the mass of the diaphragm is not negligible). The resistive term varies with frequency, except when the speed of sound in the medium is less than that in the membrane in which case the resistance is constant. In the next sections, you will see how we use these principles to physically lower the diaphragm's resonant frequency while still maintaining a greater tension on the diaphragm.

SUMMARY

We have covered all of the pertinent principles that are used in the SHOCKWAVE design without the need for extensive mathematics. Some of you, I am sure, may be familiar with these principles but it is the interrelation of all mechanical, electrical, and acoustical facets that makes SHOCKWAVE the first practical and functional device that it is.

Let's now summarize the general operating principles employed in this design. We all realize that much more bass could be attained by an electrostatic loudspeaker (ESL) if we could place the diaphragm in a box or enclosure of some sort. But as noted earlier, problems arise due to the weak forces inherent to ESLs, specifically low volume and a higher resonant frequency (plus increased "Q"). So here is where we utilize ant-noise principles to keep the diaphragms radiating in free space while simultaneously destroying the rear and out-of-phase wave form. We use a dual set of five ESL elements, in a special physical configuration and a variety of sizes, to eliminate the rear wave. This not only substantially increases the bass output, but maintains the desirable acoustical characteristics of the ESL. The added feature to this brings up the advantages that come with multiple or compound diaphragms. As we double each diaphragm, we gain a 6db increase and, furthermore, the diaphragms are doubled more than once for even more output. But what about the resonant frequency problem associated with compound diaphragms? We have solved this with a unique supporting system for the diaphragm that prohibits the increase in resonance while still enjoying the electrical and acoustical benefits of a tightly stretched membrane. You all realize that



the higher the tension on the diaphragm, mo bias voltage can be used to gain even mo potential bass output. Plus a low system reso nance translates to a potentially lower usab frequency response.

So we exploit many principles to the fullest that results in an extreme articulate but concussive subwoofer:

1. High ESL surface area for a small amount o floor space.

2. Anti-noise to eliminat most of the rearwave an gaining extreme volume leve and extended response.

3. Compound diaphragms fo increased efficiency.

4. Low resonant frequen that is the underlying neces sity of any subwoofer

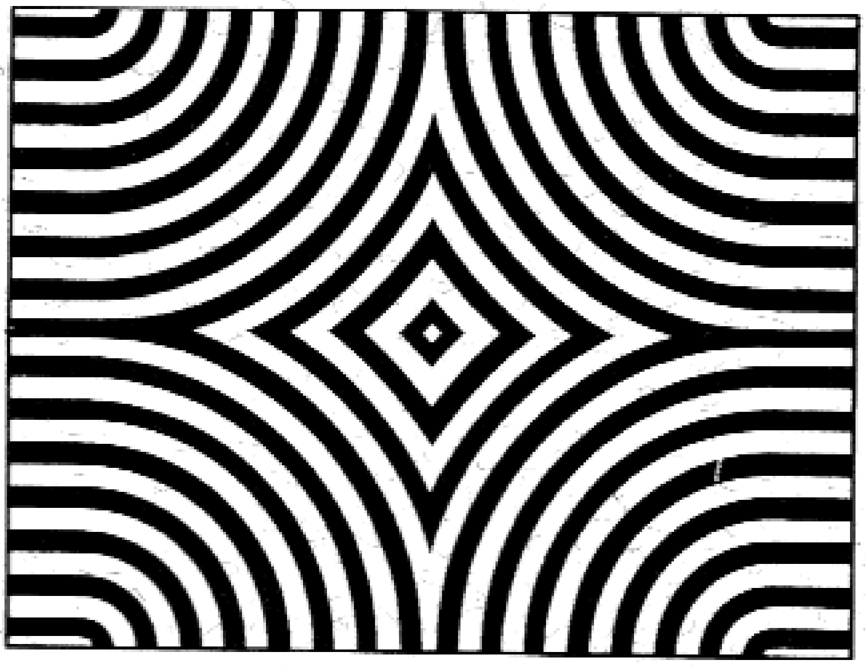
5. High bias potenti through the unique diaphragm supportive sy tem which increases both sensitivity and bas output.

6. Simple construction through the use of fl ESL elements and the respective support sy tem.

7. Low cost since the perforated metal is not critical factor and again an inexpensive su portive system for the ESL elements. The lo cost is further enhanced through the maximu use of "hardware store" materials plus very ine pensive electronics (see next section).

CONSTRUCTION

As noted earlier, we have planned this construction procedure to be as complete as possible, but we will unbiasly (no pun intended) allow the licensee to use ANY basic ESL construction method desired since every builder has their own favorite method. For those of you that are familiar with electrostatics but have no preferences in building a basic ESL element. we refer you to Audio Amateur Corp., 305 Union St., Peterborough, NH 03458. Phone:603.924.6371 Fax:603.924.9467. Audio Amateur Corp. offers several books and many past articles on ESLs and their construction. Plus they are great people to do business with!



First lets take a look at some models, sample parts, and a special set of details printed on reflective plastic. We will be referring to these during the construction procedure so you should be very familiar with them. Look in the brown envelope attached to the inside of this manuscript. It should contain:

1. One FIVE COLOR model - This is made to a scale of 2 inches = 1 foot. You will refer to this model to calculate the size of each ESL element plus how each element is mounted in relation to the other elements. It is both a front and back view of one-half of SHOCKWAVE. Consider it the left half and the right side of SHOCKWAVE is a mirror image of it. The front of it has the tallest piece and is yellow in color. The rear of it has a blue ESL element. NOTE: when I refer to <u>element</u> I am referring to a completed and mounted ESL speaker.

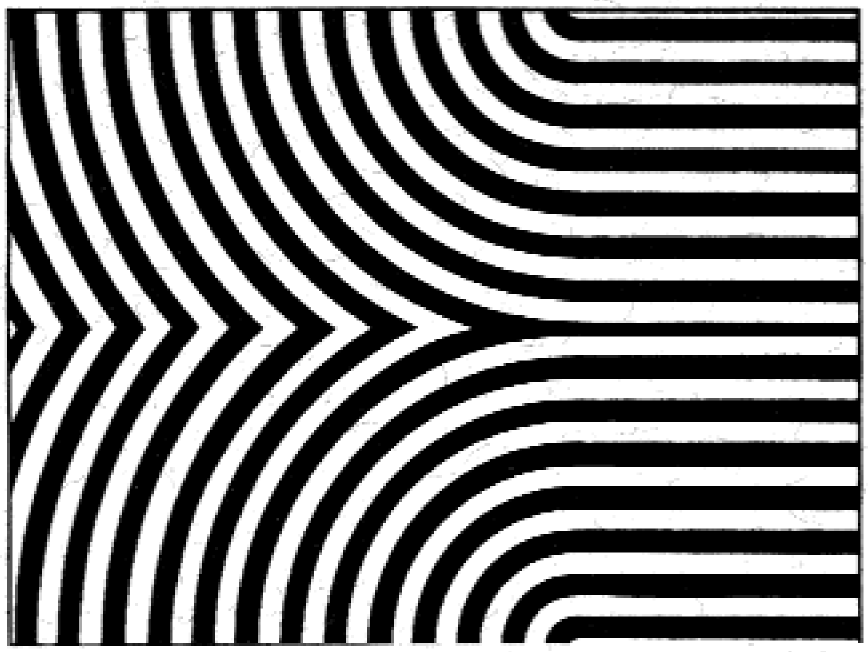
2. One METAL & PLASTIC model - This is made to a scale of 1 inch = 1 foot. It shows how a completed ESL element should be mounted. We recommend this type of PVC channel that is used in the model, but common PVC pipe may also be used. So all 10 ESL elements are mounted in this fashion and hot glue may be used to fasten the ESL to these mounting frames. Look around your local building supply for many similar plastic forms and shapes.

3. One DARK COLORED FOAM - This material is used for the spacer dots (see reflective page) and is cut into one-half inch squares and hot glued to the perforated metal and the diaphragm in the arrangement shown in figures 1, & 2. Figure 3 shows a magnified detail of the spacer dot. Use a VERY small amount of hot glue! This foam material gives SHOCKWAVE its unique tensioning and lowers the resonant frequency while still maintaining a high bias voltage to be applied to he diaphragm. You can find this inexpensive material at any hardware or building supply. It is typically used as a filter material for air conditioners. and is sometimes used in dehumidifiers as the evaporator medium. You may substitute any type of foam that has the same density and flexibility qualities as this sample.

4. One LIGHT COLORED FOAM - This sample is commonly referred to as upholstery foam. After you have completed all 10 ESLs and have them in their mounting frames, you will then begin to glue the frames to the base. A six inch square piece of this foam will be hot glued between each completed ESL, at about the center, to hold them all together. So it functions as a supportive enhancement and reduces the possibility of undesirable vibration of the sandwiched ESLs. So a total of 8 pieces will be used BETWEEN all 10 of the ESLs. None are used at the front or back of the completed unit.

Refer now to the reflective page. Figure 1 shows the spacer and spacer dot layout of the YELLOW ESL & figure 2 shows the layout for the RED ESL (see FIVE COLOR model). The ORANGE, GREEN, and BLUE elements are not shown, but it is obvious that they will look like figure 2 but will have an extra set of spacers and spacer dots added to the edge, or to put it another way, they will look like figure 1 tripled. All spacers are one-half inch wide and one-sixteenth inch thick. The DARK COLORED FOAM spacer dots are one-half inch square and one-eighth inch thick.

Figure 4 shows the top view and sequence of each ESL element (see FIVE COLOR model). Remember that the model is the LEFT HALF of Figure 4 and the RIGHT HALF is a mirror image of the LEFT HALF. The left yellow element is butted against its right yellow mate and the remaining elements are positioned according to the FIVE COLOR model. So figure 4 is a top view of the FIVE COLOR model.



Optimum spacing between each ESL element is 1. to 2.5 inches. So adjust your LIGHT COLORED FOAM thickness accordingly.

STEREO OR MONO - SHOCKWAVE can be used in either stereo or mono. If mono is desired, the unit takes the shape of a V as described. If stereo is desired, you split SHOCKWAVE into two sections both physically and electrically. The FIVE COLOR model will be the left channel and its mirror image will be the right channel. Keep each unit at a 90 degree angle to its mate and place them close to your upper range speakers. Even separated they will still perform the same as the V shaped unit. See the next page for all particulars on wiring them and carefully examine the schematic diagrams provided.

BASE & TRIM - The base must be made out of any plastic. PVC or Plexiglas work very well and need not be very thick since it will be hot glued to a piece of one-half inch particle board. Even one-sixteenth inch is fine for the plastic base. All ESL elements will be glued to the plastic base that is then glued to the particle board. You may then hot glue this assembly to a patio stone concrete base, but this is optional. Apply a loose mesh grill cloth to the entire structure in a SOCK shape. You can then add wood or metal trim as in the photo of SHOCKWAVE.
NOTE: Be sure to cover all high voltage parts with a grill or grill cloth for safety - VERY HIGH VOLTAGES ARE PRESENT!

NON-ELECTRONIC MATERIAIS AND PARTS LIST - Refer to the FIVE COLOR model to obtain quantities and sizes for the completed SHOCKWAVE.

1. Perforated metal - Use the hardware store variety that is 2' X 3' and is termed LINCAINE DECORATIVE METAL. It is easy to find and inexpensive. The silver colored works the best.

2. Spacer material - Your preference.

3. Spacer Dot material - DARK COLORED FOAM.

4. Mylar film - See our catalog or your favorite plastics supplier: one-third to one-half mil thickness.

5. Graphite - Hardware store lock graphite.

6. Upholstery Foam - Sewing and craft supply shops.

7. PVC pipe, channels, plastic base and particleboard - Building supply outlets.

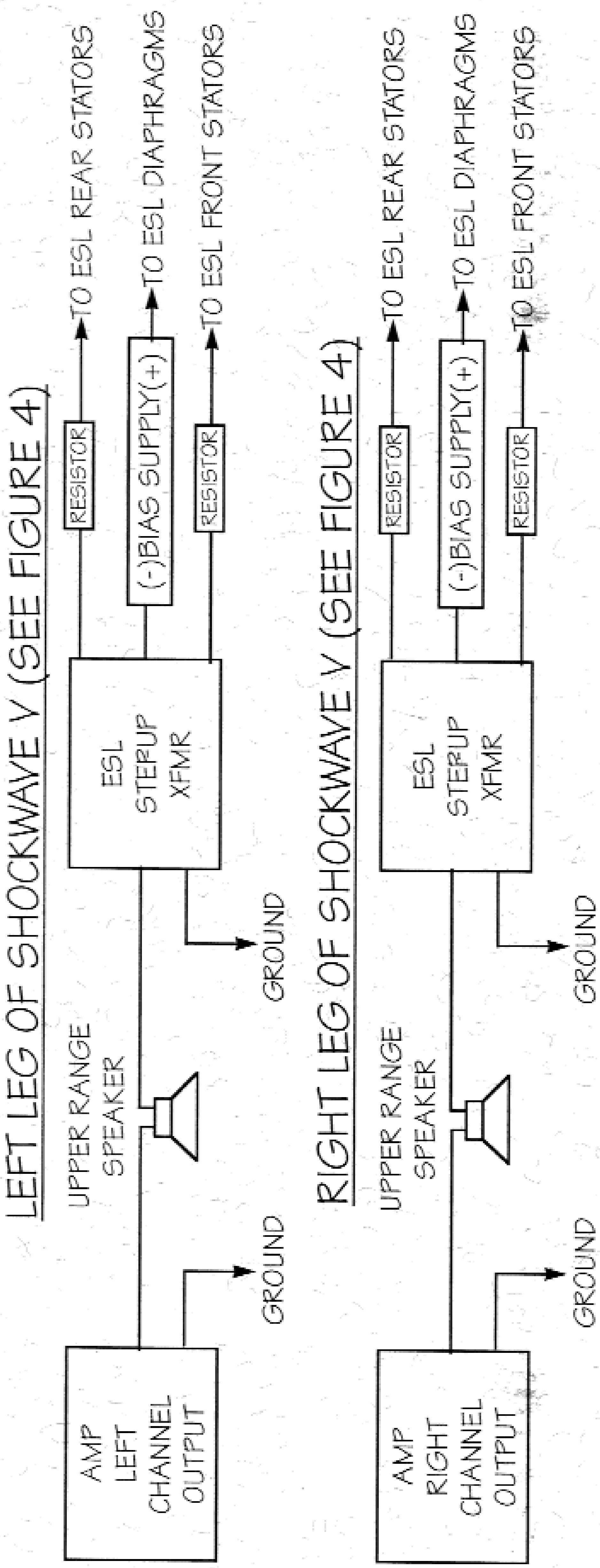
8. Grill Cloth - Sewing supply stores.

ELECTRONICS & WIRING - There are very few electronic parts required for SHOCKWAVE and the crossover point is set with a simple one-half watt resistor (Radio Shack). The resistor value is affected by the specs of the step-up transformer, but values between 500.k ohms and 3. meg ohms will generally be the correct range. The main components required are two vacuum tube output transformers and one or two 2K. to 5.k VDC bias supplies. The output transformers can be salvaged from old vacuum tube amplifiers and the bias supply from salvaged Xerox machines. As per the wiring diagram, SHOCKWAVE is wired IN SERIES with your upper range speakers whether you are using it as a stereo or mono subwoofer. As noted earlier, stereo subwoofing is attained by PHYSICALLY separating each leg of the V (see figure 4 and the FIVE COLOR model). All ESL elements are wired in PARALLEL and this includes the diaphragms. Although two bias supplies are shown, only one is really needed and its positive output would go to ALL diaphragms and its negative output will go to the center tap of BOTH step-up transformers.

CONSTRUCTION STEPS:

1. Use the FIVE COLOR model and the reflective page to construct 10 ESL elements - use favorite method.
2. Cut PVC pipe or channel to frame each ESL element using hot glue (see metal & plastic model).
3. BASE: Cut a sheet of plastic and particle board to a size that conforms with the size and spacing of ALL COMBINED legs of SHOCKWAVE (see figure 4 and your finished ESLs to determine size). Sandwich together using hot glue.
4. Use hot glue or plastic cement to attach all ten ESL elements to the base. Be sure to add the upholstery foam between each element.
5. Wire all ESLs according to the diagram at right.
6. Cover all elements with grill cloth and apply the trim of your choice if desired.

7. BLAST OFF!



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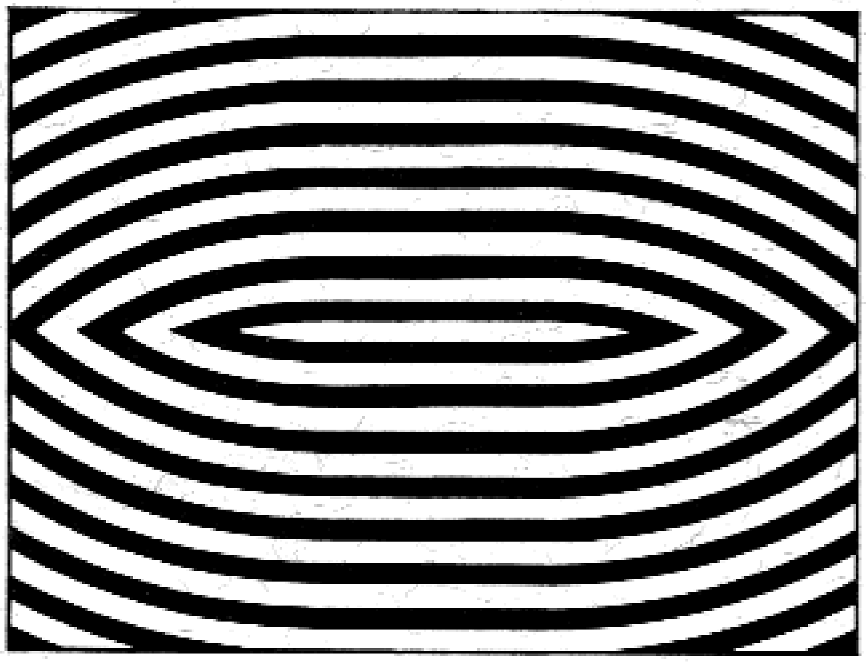
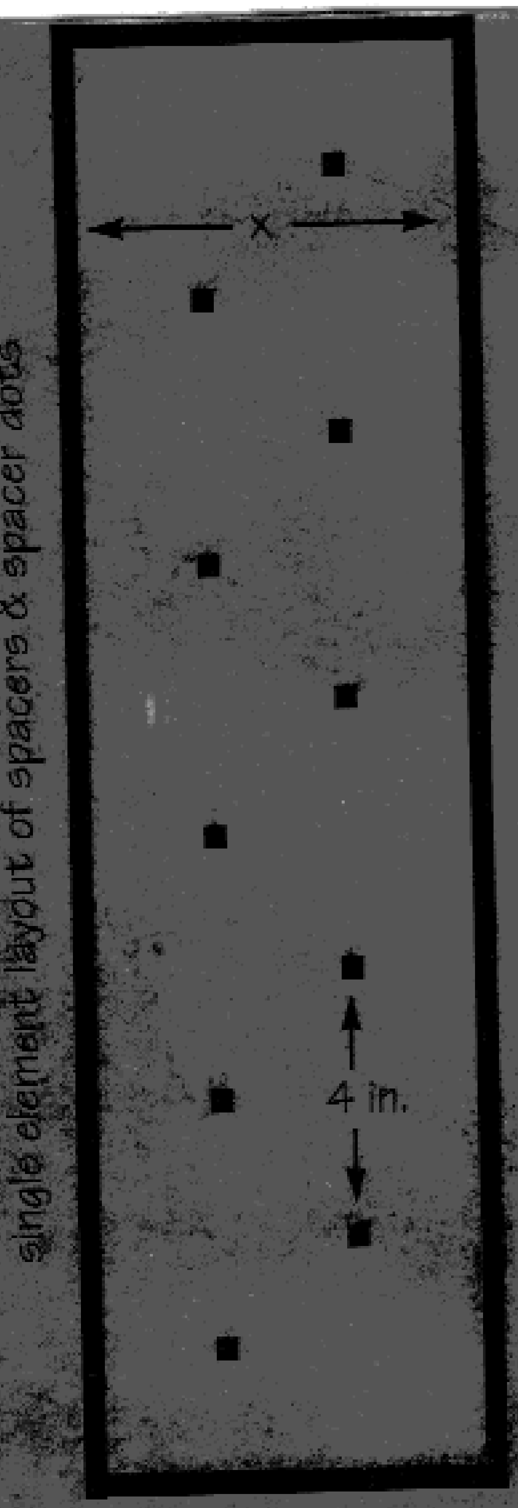


figure 1



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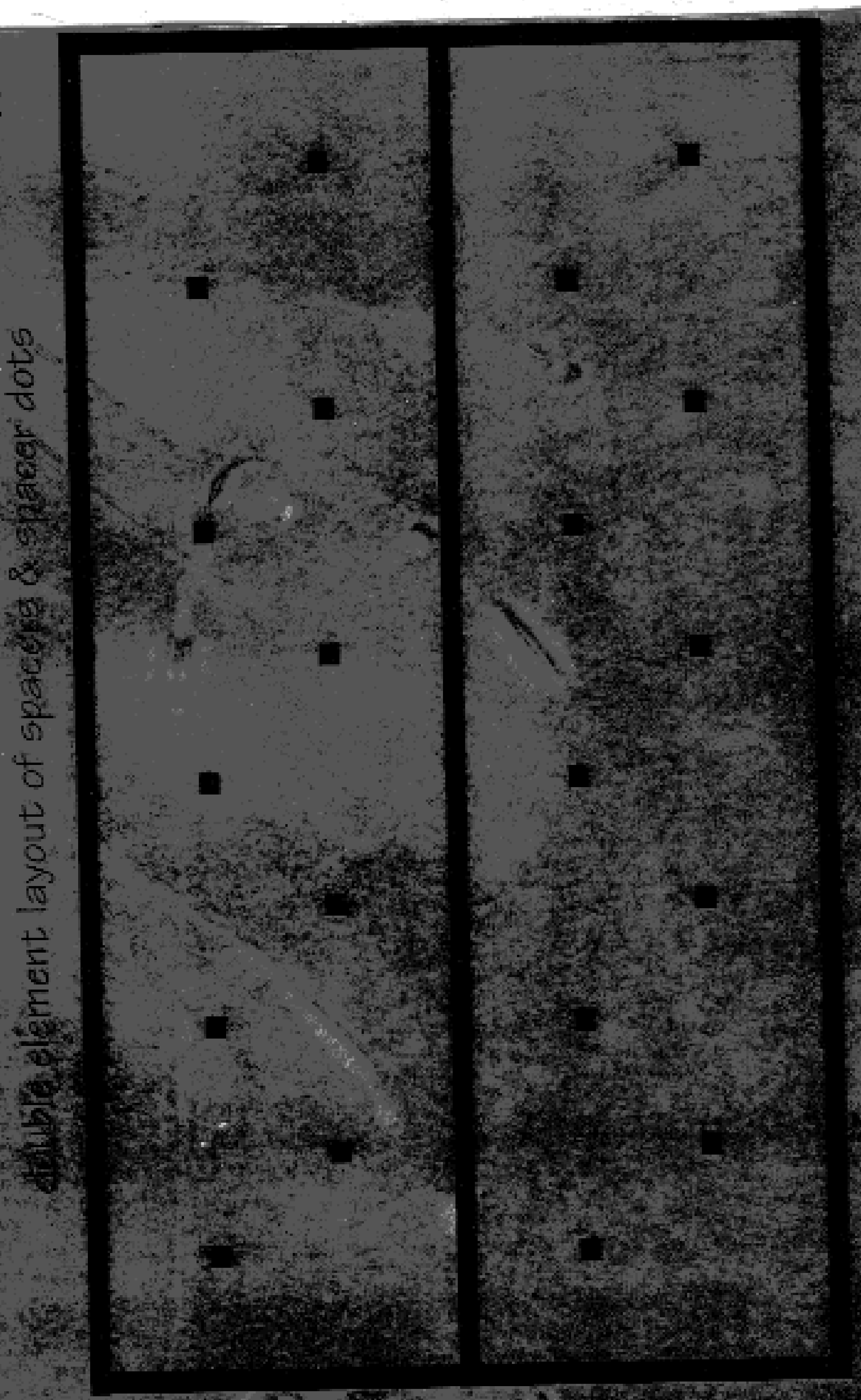

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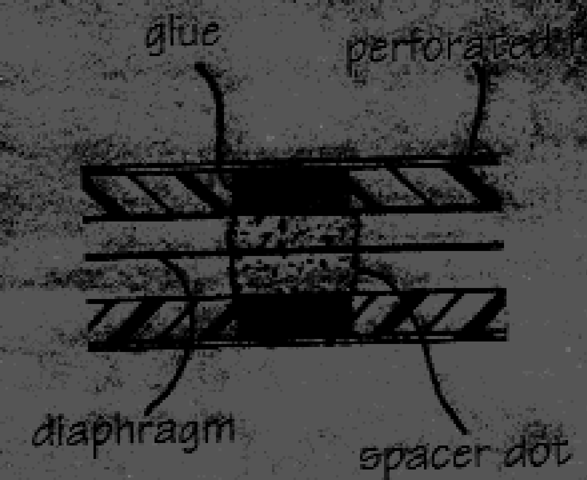
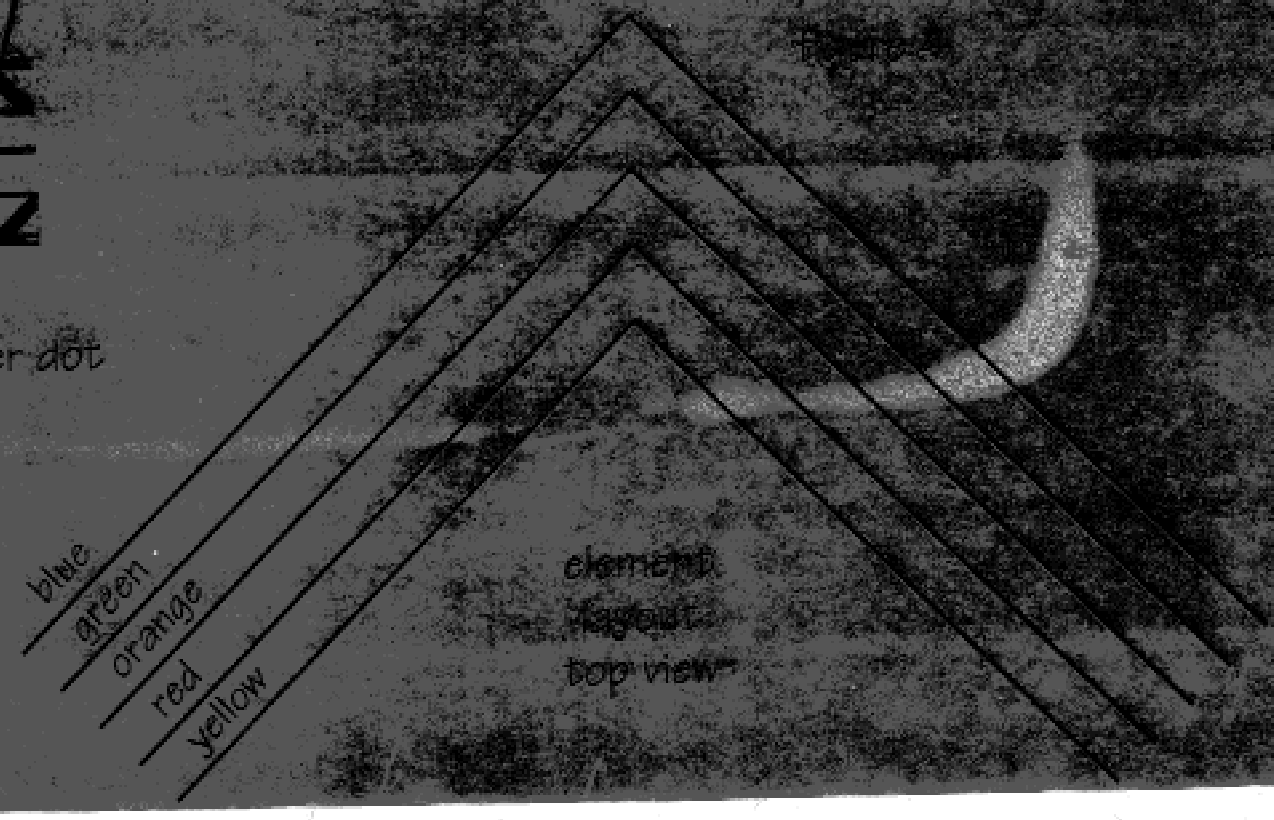


figure 3



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