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ON THE DESIGN, MEASUREMENT, AND EVALUATION OF LOUDSPEAKERS

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INTRODUCTION AND BACKGROUND

At first it may seem rather strange to be discussing basic problems of design and evaluation of a device that has been in use for many decades and for which analysis and design is discussed in almost every text on applied acoustics. The very fact that engineers are continually striving to improve the loudspeaker would almost naturally imply the existence of a measurement and evaluation criterion by which they meter their progress.

However, when we look more closely into the problem we find that present measurement techniques fall far short of reaching the goal of complete objective evaluation of speakers. This is not surprising because to obtain a complete set of objective parameters for loudspeaker measurement would require a knowledge of a sufficient set of physical variables and their correlation to hearing to not only characterize all sounds that we can distinguish but to assign a quantitative measure to the magnitude of the differences between these sounds. It is recognized by all serious researchers of the auditory process that this goal is still remote. At present we have, as examples, an approximate notion of what fluctuations in the magnitude of the Fourier spectrum of pressure are detectable on complex signals like speech and music, very limited knowledge about detectable phase variations, and almost no quantitative knowledge concerning the subjective effects of spatial aspects of complex signals incident upon a listener.

The problems that face us in the quest for objective measures for speaker performance can be broadly divided into two classes. In the one class we have the problems of correlating the existing quantitative measures with the subjective experience of listening to music or speech. This correlation is necessary both to establish more accurate performance criteria and to avoid costly overdesign which brings no benefits to the listener. In the other class we have the problems of searching for additional objective measures that are necessary for meaningful characterization of the performance of loudspeakers. From recent research it is becoming clear, for example, that the ratio of direct-to-reflected sound radiated by a speaker into a room and the effect of this ratio upon the spatial characteristics of the sound perceived in the room are highly correlated to the quality of the sound reproduction.

In addition to the problem of determining appropriate physical variables and their correlation to hearing we face the basic tasks of determining objectives and recognizing constraints that may prevent their full realization. It would perhaps be possible to achieve a consensus on the objective of providing the home music listener with the same auditory sensation that he would receive at the live performance. If we did not pause to consider some practical constraints we would immediately move in the direction of a large number of channels with an even larger number of speaker systems and time delays, all installed in an anechoic environment. Very effective experiments of this type have been conducted for research purposes. An excellent demonstration of such a system is provided by the Laboratories of Philips Gloielampenfabrieken in Eindhoven, Netherlands. However, if we introduce the practical constraints that limit us to two channels and to rooms the size of those found in average homes, it is safe to say that sound reproduction is at best a compromise with numerous shortcomings. The instant that we admit that the reproduction will not exactly meet the desired objective then we admit the existence of an error and we are obliged to determine a measure for this error if we are to evaluate one system relative to another. For the reasons discussed earlier, the measure for this error still resides in the domain of subjectivity, shrouded with all the complexities of personal value judgments.

Confronted with this situation, the engineer has attempted to extricate himself by applying objective criteria that have their origin in the well established facsimile conditions for linear systems. The justification for the use of these conditions has been based upon the fact that the first-order model of a loudspeaker is linear and that nonlinearities can be handled as perturbations on the linear model. However, direct application of the facsimile conditions to loudspeaker measurements presents problems that are far more basic than the issue of linearity. To illustrate this let us consider, for example, the parameter of frequency response, which is the parameter that has received the most attention in loudspeaker design over the past three decades. In system theory one defines a system in terms of variables at its input and output terminals. If the input to a linear system is $x(t)$ and the output is $y(t)$ then the system can be characterized to either its complex system function $H(\omega)$ (where ω is the radian frequency) or, equivalently, by its impulse response or Green's function $h(t)$. The facsimile conditions (those for which $y(t)$ differs from $x(t)$ by at most a time delay) require that the magnitude of $H(\omega)$ be constant over the frequency. Generally for music or speech applications only the magnitude criterion is applied on the hypothesis that phase irregularities are not observable in the reproduction of music or speech. This hypothesis has been substantiated to a considerable degree by psychoacoustic experiments involving limited phase changes in the frequency range above 200 Hz. It is open to considerable controversy for lower frequencies. In any case, we shall list some of the problems that arise if we try just to apply the magnitude criterion to the measurement and evaluation of loudspeakers.

1. The loudspeaker input is well defined by its electrical terminals. Where is its output?

2. Can the output be considered to be located at a point in a room without including the room in the system function and predetermining the position of the listener in the room?

3. Even if we can define an output, how do we measure a loudspeaker designed to operate in a room? The room interferes with the measurement and the loudspeaker radiation changes if we try to measure it in a free field.

4. Even if we could solve the previous problems and could measure some appropriate frequency response, how would we evaluate one speaker relative to another without an exhaustive study of the correlation between the measured variable and its subjective effects? This is not as simple as it may seem because peaks and dips of a given size are much more noticeable in some portions of the spectrum than in others.

5. Last, but by no means least, how would we know when to stop optimizing a loudspeaker with respect to its frequency response? The gross irregularities in the frequency characteristics of the room caused by normal modes mask certain types of speaker irregularities. How would we determine if this is the case in a given situation?

A simple list like that above serves to alert us to examine very carefully those concepts that we have taken for granted because they have been used for such a long time or because they relate to other situations in which they clearly apply. With this as background, let us begin our investigation of loudspeaker design and evaluation by examining the sound fields through which we perceive music.

SOUND FIELD CONSIDERATIONS IN MUSIC REPRODUCTION

Conventional loudspeaker design has tended to optimize parameters that are appropriate for the case in which the direct radiated field from the loudspeaker is predominant at the position of the listener. The emphasis upon flat frequency response on axis in free field and upon improving the dispersion at high frequencies are examples of this design. This approach might be appropriate if the speakers are to be listened to out of doors or in an anechoic chamber. However, it leaves much to be desired when the listening environment is the living room and the object is to simulate as many as possible of the properties of the sound in a concert hall. In order to appreciate the reasons behind this observation let us first consider the nature of the sound fields that are present in concert halls. Then we can relate this knowledge to the design and evaluation of speakers to perform in living rooms.

For purposes of illustration we shall consider a spherical source in various environments. If the source is located in an anechoic environment, direct solution of the wave equation in spherical co-ordinates enables us to express the pressure $p(r,t)$ at any point a distance r from the center of the sinusoidally excited sperhical source (outside the source) by the equation

$$p(r,t) = R_e \frac{A}{r} e^{(\omega t - kr)} , \qquad \text{Eq. 1}$$

where the constant A is the complex amplitude of the pressure at a unit distance from the center of the sphere, ω is the radian frequency of the source, t is time, k is the wave number, and R_e denotes the operation of taking the real part.

The factor of interest to us in this expression is the $\frac{1}{r}$ dependence of the amplitude of the pressure. It points out that in an environment in which reflecting boundaries are absent (i.e., in which the direct field is dominant) the sound pressure will drop as $\frac{1}{r}$ with increasing distance from the source. In terms of sound pressure level (S.P.L.) measured in db, we have the familiar result that the S.P.L. decreases by 6 db. for each doubling of the distance from the source.

Now let us consider the situation that prevails when a loudspeaker is brought into a room. Equilibrium conditions are established, from consideration of conservation of energy, when the average power radiated by the source is equal to the average power absorbed by the room. The consequences of this simple statement are best appreciated if one initially considers the limiting case of a room with perfectly reflecting walls and no sound absorption by the air. In this case the pressure in the room would continue to build up until all the radiation from the loudspeaker was stopped by the back pressure. In this limiting case, we would have no direct field of the loudspeaker present in the room, and we would say that the reverberant field comprised the total field. (A sound field is said to be direct if it has not undergone any reflections. The reverberant sound field in a room is the sound field comprised of reflected energy). There are several observations we should make about this reverberant field for use later in our discussion. First of all, the pressure fluctuations as we move about the room are due only to the standing waves of the room and bear no relation to our proximity to the loudspeaker. Secondly, the Fourier spectrum of the pressure measured at any point in the room is related to the total energy that the speaker had radiated and not to its radiation along any axis. Finally, the angles of incidence of the sound waves at any point in the room are widely distributed, as opposed to the highly directional incidence that would result from the same sound source if placed in an anechoic environment.

Before drawing any conclusions from the observations made in the extremes of anechoic and totally reflecting environments, let us consider the environment of the concert hall which lies between these extremes and has some of the properties of each. For ease of calculation let us consider a spherical source in the center of a large irregular room. This idealized case will enable us to algebraically express results which, as can be verified by measurement, will indicate the approximate behavior of the sound fields in concert halls and in living rooms. For the case under consideration the S.P.L. can be expressed as

$$\text{S.P.L.} = B + 10 \log_{10} \left(\frac{1}{4\pi r^2} + \frac{4}{R} \right) \qquad \text{Eq. 2}$$

where B is a constant and r is the distance from the center of the source. The symbol R denotes the room constant which depends upon the area of the room surfaces, the average absorbtion coefficient and, if air absorbtion is included, the volume of the room.*

*See Acoustics, L. Beranek, McGraw-Hill, 1954, for a detailed treatment of this topic.

The room constant R increases with the size of the room and with the average absorption coefficient.

As a check on Eq. 2 consider the case of a room of infinite size. In this case, the room constant R becomes infinite and Eq. 2 reduces to the case discussed earlier of a sherical source in an anechoic environment. Note that the S.P.L. decreases by 6 db. for each doubling of the distance from the center of the source as in the anechoic case.

Now let us examine the effect of the room on the S.P.L. in Eq. 2. When $1/4\pi r^2$ becomes less than $4/R$ the S.P.L. approaches a constant with increasing distance r from the center of the source and the reverberant field is dominant. For smaller r the first term in the brackets is larger than the second and the direct field dominates. A curve of S.P.L. vs. r from Eq. 2 is plotted in Figure 1. Even though the curve of Fig. 1 was obtained under the assumption of a large irregular room, it predicts the approximate behavior for a wide variety of rooms as is easily verified by measurement. In general, the S.P.L. drops off as the distance from the source increases until the direct field becomes smaller than the reverberant field. Beyond this point the intensity is independent of distance and its variation with room position is a function only of the standing wave pattern in the room. In fact, if one-third octave noise is used, then, particularly at the higher frequencies, the reverberant field can be quite constant with position.

The relevance of Fig. 1 to loudspeaker design and evaluation becomes evident when we examine the sound field in concert halls and find that for virtually all seats the reverberant field is dominant. Even for a large hall such as Symphony Hall in Boston the room constant is in the order of 18,000 square feet, resulting in the reverberant field equaling the direct field at about 19 feet from the source. Recall now one of our previous observations regarding the reverberant field. In particular, since the energy in this field arrives at any point via reflections from the surfaces of the room, the angles of incidence of the arriving sound energy are widely distributed. This spatial property of the sound incident upon a listener is a parameter ranking in importance with the frequency spectrum of the incident energy for the subjective appreciation of music. (In a related portion of the research, not reported in this paper, experiments were conducted in which only the spatial characteristics of the incident sound were altered, without affecting its frequency spectrum. The experiments conclusively demonstrated the importance of these spatial characteristics.)

Let us now relate our findings concerning the nature of the sound field in concert halls to the design and evaluation of loudspeakers. The principal effect they have is to change the present design and measurement techniques, which are based upon direct field considerations, to new techniques based upon a combination of direct and reflected sound produced in the listening environment. The wide distribution of the angles of incident sound energy that prevails in concert halls, and in most other environments where music is performed, is a salient factor that can, to a limited extent, be simulated in a living room.* Conventional speaker design however results in the dominance of the direct field from the loudspeakers with the consequent localization of stereo sound in two points and the noticeable lack of fullness or openness of the reproduced sound. It is interesting to note that even the efforts to design speakers with better high frequency dispersion are based upon direct field concepts. The object of improving the polar patterns of tweeters is to make the direct field, which is the dominant field in listening to conventional speakers, more uniform as we move around the speaker. However, even if perfect dispertion were achieved, as long as the direct field is dominant the listener would receive the majority of sound energy from the direction of the source and would not experience the wide dispersion of incident angles that is so important in the appreciation of live music.

The requirement of a sufficient ratio of reverberant-to-direct sound produced by a speaker in a listening room is only one of several spatial problems that face the loudspeaker designer. Another important spatial factor is pertinent to transients that occur in times short compared to the build up time of the reverberant field. It is well known that one's ability to sense the direction or extent of a source sound is based upon transient rather than steady state signals. An easily performed experiment will serve to illustrate this point. Place a number of sources across the front of a reverberant room and let them simultaneously produce a sine wave of the same frequency. An observer sitting in the room will generally not be able to detect whether one or all of the speakers are producing the sine wave. However, if the sine wave is momentarily interrupted or if any sudden transients are introduced through the speakers, the observer will immediately be aware of which speakers are producing the sound. This result is not new and it is not surprising since we sense the direction of incident waves largely by difference in the time of arrival of the waves at both ears. If the reverberant field is dominant, sound energy is incident upon the observer from a wide distribution of angles with no particular preference for the direction of the source and we would not expect to be able to locate the source. On the other hand, for a transient, there is a time interval during which the predominant incident energy containing the transient is that of the first waves to arrive from the vicinity of the source. This provides the information regarding the location and extent of the source.

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*The wide dispersing of angles can be simulated but, of course, the long time delays associated with the concert hall reverberant field cannot be obtained without the use of additional speakers and delay units.

Let us now attempt to integrate both of the spatial considerations that we have discussed into a practical speaker design and see what constraints they impose on the design. The first consideration, that on the dominance of the reverberant field, when considered in the light of the absorption coefficients and the moderate dimensions of home listening rooms, imposes the constraint that the speaker should be placed out in front of a wall and should radiate the majority of its energy away from the listening area toward the walls. In this way it is possible to increase the ratio of the reverberant-to-direct fields in the listening area as required. The second consideration, relative to the listener's perception of the location and extent of the apparent sound source, dictates that the energy radiated from the back of the speaker should be directed toward the wall at selected angles to yield an apparent source that is widely distributed along the walls in much the same sense that an orchestra is distributed across the stage. The solution that we evolved to best meet these constraints is indicated schematically in Fig. 2. It is a pentagonal shaped structure that radiates 89% of its energy from the two back panels and 11% of its energy from the front panel. It provides the necessary ratio of reverberant-to-direct sound and with the rear faces of the enclosure angled at about 30 degrees to the wall, the wavefronts emerging from the first reflections from the back wall and the second reflections from the side walls produce an effective source that is much larger than the actual enclosure and that is well distributed across the wall.*

As is apparent, we are attempting to apply our knowledge of sound fields in concert halls and listening rooms to impose constraints upon the design of a loudspeaker at as broad a level as possible. Up to this point, we have constrained only the spatial characteristics of the radiation. We have not yet discussed the frequency spectrum of the radiation or the problems of designing the transducers to produce the desired spectrum without audible coloration of the sound. We shall now take up these topics and examine their implications for both the design and measurement of loudspeakers.

FREQUENCY RESPONSE

Earlier in this paper we raised some questions regarding the basic meaning of the frequency response of a loudspeaker. At this point it might appear that the constraints that we have now introduced on the spatial radiation of a loudspeaker will only further complicate the problem. For example, uniform polar response and flat frequency response on an axis are incompatible with these constraints. Fortunately, however, these spatial constraints simplify rather than complicate the problem of obtaining a meaningful definition of frequency response. As we recall, the constraints

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* The dispersion of the transducers in the vertical plane was regarded as sufficient not to require vertical angling of the rear panels.

dictate that the speaker radiation should result in a dominant reverberant field in the listening area. For this field it is the frequency spectrum of the total energy radiated by the speaker that is significant rather than the frequency response along any axis. The question naturally arises concerning what the frequency response of the total radiated energy should be since, certainly, the room will absorb and reflict different frequencies differently and cause the spectrum of the reverberant energy to differ from the spectrum of the radiated energy. It is readily appreciated that variations caused by the listening environment are common to all live performances and to all loudspeakers. Under the circumstances it is reasonable to adopt the criterion that when the loudspeaker is properly placed relative to the rear reflecting wall, the frequency response measured with respect to the total radiated acoustical energy should be flat. This simply stated criterion poses a difficult and interesting measurement problem, however. The reflecting wall behind the speaker influences its radiation and must be present during measurements. Conceivably the speaker could be measured in front of a large plane in free space. However, with the use of only pressure measuring equipment, the measurements must be made far enough away from the speaker to assume the usual relation between pressure and power radiation in a plane wave. In addition, the measurements must be made over an entire surface inclosing the radiation of the speaker. An alternate approach is to place the speaker in a room and measure the narrow band noise spectrum of the reverberant field. This measurement will suffer from irregularities in the spectrum caused by standing waves and by the frequency dependent absorption coefficients of the room. For spherically shaped speakers it is possible to accurately calibrate what the room does to the source and then to subtract its effect from the reverberant field measurements to determine the spectrum of the total radiated energy. For speakers of other than spherical shape we presently have no way of accurately calibrating the room. In this case we resort to an approximate technique of calibration by a spherical source coupled with spatial averaging in the reverberant field to determine the spectrum of the total energy radiated from a speaker.

THE CONCEPT OF DIFFERENCE EXPERIMENTS IN LOUDSPEAKER DESIGN AND EVALUATION

Thus far in this paper we have only established spatial and spectral design objectives for high fidelity loudspeakers and we have not considered the many ways in which the practical transducers may color the sound reproduction. The portion of the research program directed toward the latter problems occupied over four years of study, the details of which will be published elsewhere. However, the research is sufficiently relevant to the topic of this paper to at least mention the basic experiment and summarize the principal results.

This portion of the research addressed the problem of speaker system design with the object of eliminating audible colorations of the sound due to transducer imperfections. As part of the study it was necessary to address the very basic question -- how do we determine if we have eliminated all these colorations?

It was observed at the start of this paper that we are a long way from obtaining a model of the hearing process that is sufficient to objectively characterize all sounds that can be subjectively distinguished. As long as this situation prevails we cannot escape the conclusion that the ultimate evaluation of a loudspeaker system is subjective. However, this is no reason to throw up our hands and conclude that loudspeaker design must be guided in the fashion of a random walk by a myriad of individual biases and preferences. On the contrary, there are scientific experiments that can be conducted which make use of, and provide us with information about the subjective hearing process but eliminate the aspect of personal value judgment. The key to these experiments is that the subject must only be asked to detect the possible existence of a difference between two samples of sound and not asked for any evaluation of the difference, should it exist. Two such experiments will be mentioned in the remainder of this paper and their implications for speaker design and evaluation will be summarized.

DIFFERENCE EXPERIMENT TO TEST FOR AUDIBLE COLORATION OF SOUND IN AN ARRAY OF SMALL SPEAKERS

The first experiment used a digital computer to produce a tape of selected music and speech samples as it would have been recorded in a living room if the samples were played through a hypothetical one-eighth of an ideal pulsating sphere placed in a corner. The same music and speech samples were also played in the same room through an approximation to the ideal sphere consisting of 22 four inch speakers placed on an octant of a sphere of 20 inch radius driven by a computer derived electronic equalizing network. The two tapes were subjectively compared and all observers were unable, in an A-B test, to distinguish the computer processed music and speech from the speaker processed music and speech.

The details of this experiment will appear elsewhere, however the basic block diagram of the experiment is shown in Fig. 3. The tape of the sound from the ideal sphere was obtained from measurement of the Green's function of the room by means of a spark discharge, and subsequent digital convolution with the music and speech samples. The object of this paper is rather to illustrate the practical implications of such difference experiments.

The experiment proved that, with proper frequency equalization, the multiplicity of closely spaced small speakers on the spherical surface can produce music and speech signals in a normal listening environment that are subjectively indistinguishable from those that would be produced by an ideal pulsating sphere in the same environment having no resonances, phase shift, diffraction, or distortion of any kind. It constructively demonstrated that the resonances and distortions of the individual cones in the spherical array of speakers were not audible. It also provided a design alternative to eliminate the sound coloration caused by resonances of speaker systems using only a small number of speakers and by irregularities in the radiated energy spectrum of systems employing crossover networks. As we would expect, there are many measurable differences between the signals on the tape containing the computer processed music and the tape containing the recording of the spherical array of small speakers. The significance of this difference experiment is that it enables us to draw basic conclusions relative to loudspeaker design which include the ultimate subjective evaluation and yet avoid the problems of value judgment. (The reasons why a multiplicity of small speakers with electronic equalization can achieve the same subjective quality of music reproduction as that produced by an ideal surface involve both the distribution in frequency of the resonances in different cones and the dominance of the normal mode structure of average listening rooms.)

DISTORTION DETECTION BY DIFFERENCE EXPERIMENTS

If we make a model of a loudspeaker that includes its sources of nonlinearity we find that in all cases there are frequency dependent transfer functions, that are also functions of the environment, in the link between the source of the nonlinearity and the ear of the listener. The radiation impedance and the room are examples. These frequency dependent transfer functions drastically affect the measurable distortion since they control the balance of harmonics to fundamental for a sine wave test signal. Even the anechoic chamber does not solve the distortion measurement problem because the spectrum of the loudspeaker radiation changes from that in a room -- particularly in the low frequency range where distortion figures are the largest. Couple these problems with the fact that we have n no satisfactory correlation between any of these loudspeaker distortion measurements and their subjective effects on music reproduction and we realize the need for a method of detecting distortion in a way that is meaningful to the listener in the environment for which the speaker is intended.

The second difference experiment that we shall present has been constructed to determine the presence of audible distortion in speaker systems of any design and to determine at what level of radiation the distortion becomes audible.

All the common forms of loudspeaker distortion in well designed systems share the property that they are increasing functions of intensity of radiation with negligible distortion at very small intensity. A difference experiment is depicted in Fig. 4 that makes use of this fact to ditermine the level of intensity at which a speaker begins to generate audible distortion in music reproduction. Selections of music and speech are played through the speaker in a listening room a number of times at successively increasing intensity levels that are monitored in the room. Binaural recordings of each successive speaker playing are made at a constant, standard level. The various recordings are synchronized on parallel tracks of an eight channel tape recorder. The subject is then given an A-B presentation of the sample that was recorded with the speaker playing at the lowest level and a sample representing a higher speaker level. Both samples are, of course, presented at the same level to the subject and he is asked only to try to detect a difference. The level of intensity produced by the loudspeaker for which a difference is first detected is then a measure of the performance of the loudspeaker. This measure is pertinent to the ultimate subjective evaluation but it is obtained without introducing the problems of individual value judgment or a priori notions of the sound of distortion. In this experiment, care must be taken to include a variety of source materials for which non-linear distortions would be most audible. The experiment is still in the development phase and more quantitative data will be available later. Preliminary results indicate that for most high fidelity loudspeakers, audible nonlinear distortion on music or speech is definitely one of the more minor of their shortcomings, notwithstanding the fact that distortion measurements, depending upon the technique used, can be quite large.

As illustrated by the two examples above, the virtue of the difference experiment is that it gets right to the issue of what is audible and by-passes the measurement of numbers whose values have not been satisfactorily correlated to what is heard when listening to music over loudspeakers. This type of experiment can, of course, also be used to correlate objective measures with the subjective perception of sound. It is hoped that in time, objective measures with sufficient correlation to hearing will be developed to replace the present need to involve subjects in the difference tests.

CONCLUSION

The results of the research program reported on in this paper are best summarized in outline form.

1. We have a better understanding of the roles of the direct and reverberant fields in the reproduction of music in the home.

2. An approach has been developed for certain testing and evaluation of loudspeakers that makes use of the subjective process of hearing and excludes value judgment.

3. As a consequence of the above research, we have learned several design constraints that result in better reproduction of music in the home. They are:

   a) The radiation of 80 to 90 percent of the total energy from the speaker at selected angles toward the wall behind it.

   b) For speakers designed to provide a proper ratio of reflected-to-direct sound, the criterion of flat frequency response of total radiated engergy replaces the criterion of flat frequency response on axis or flat frequency response with respect to the average spectrum measured in a room.

   c) The use of a multiplicity of closely spaced loudspeakers with full range electronic equalization can overcome the problems of sound coloration introduced by resonances in systems with a small number of transducers and by irregularities in the output of systems employing crossover networks.

The work described in this paper has hopefully brought us a little closer to our goal of understanding and solving the problems of music reproduction in the home. However, let us have no illusions that we can in any practical way create the sensation of the concert hall in the living room. There is still a long way to go and many challenging questions of practical importance to be answered. If the work described in this paper serves to stimulate the type of research that will advance us from the alchemy of cone doping and port tuning toward a basic understanding of the problems we face in producing better music in the home, it will be a source of satisfaction to the many people who have contributed to this project over the past twelve years.

ACKNOWLEDGMENT

Over the twelve years of this research many people have made substantial contributions in the performance of the experiments and in the generation of valuable ideas and criticisms. The basic computer experiment, simulating the ideal transducer, was made practical only through the techniques developed by Prof. T. Stockham to reduce the time of convolution computations. Professors J. Bruce and D. Nelsen contributed to the instrumentation of this experiment and J. Wawzonek and D. Steele, along with graduate and undergraduate students, prepared experiments and analyzed data.

It is with great appreciation that we acknowledge the cooperation and assistance of Dr. Erich Leinsdorf, T. Perry, and the members and management

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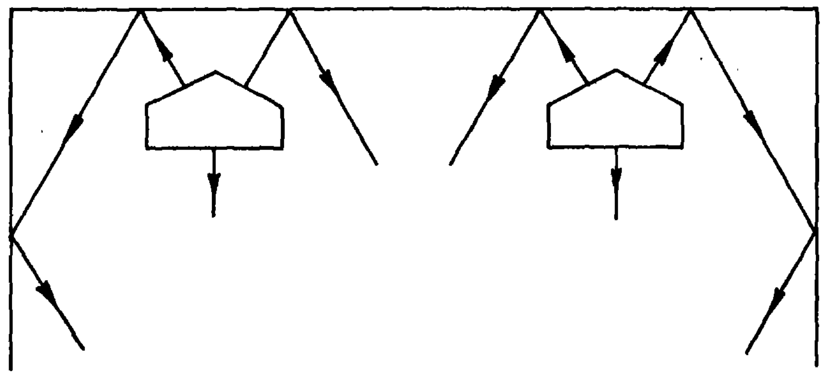
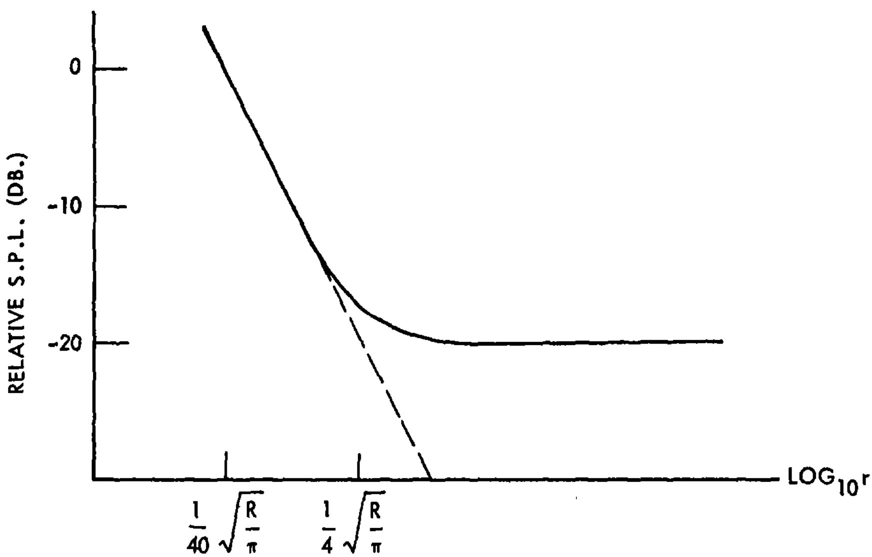


FIGURE 2

Pentagonal shaped loudspeaker enclosures designed to meet the two spatial constraints discussed in the text.



DISTANCE FROM ACOUSTIC CENTER OF A SPHERICAL SOURCE

FIGURE 1

S.P.L. vs. distance from an omnidirectional source located in the center of a large irregularly shaped room.

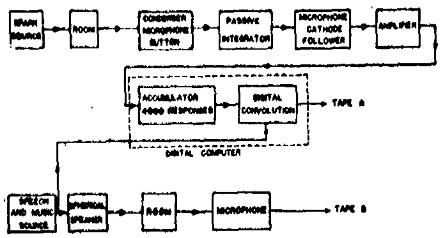


FIGURE 3

Difference experiment to study array of small loudspeakers

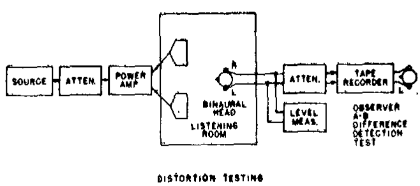


FIGURE 4

Difference experiment for distortion testing.