Qui di seguito sono riportati I due articoli di Amar G. Bose sul sistema Direct/Reflecting implementato sulla serie 901- I testi sono stati convertiti con OCR e non sono ancora perfettamente corretti.

I nuovi diffusori Opera che utilizzano il CLD e la Tripletta si basano su studi come questo. Rispetto ai diffusori sviluppati da Bose il sistema Opera presenta tre importanti differenze:

- **la sorgente che irradia il suono posteriormente, alimentando il campo riflesso, è realizzata con dipoli in modo da non interferire con il suono diretto**
- **la sorgente posteriore irradia da 2000 Hz in su (e non su tutto lo spettro)**
- **il livello della sorgente posteriore è circa uguale o inferiore a quello della sorgente frontale (mentre Bose usa un rapporto 8 a 1).**

Amar G. Bose Professor of Electrical Engineering M.I.T.

Sound Recording and Reproduction Part One: Devices, Measurements, and Perception

Amar G. Bose received his S.B. and M.S. (1952) and his Ph.D. (1956) from M.I.T.; his doctoral thesis was on the characteriration of non-linear systems, a matter under investigation by Professor Norbert Wiener. Professor Bose is co-author of *Introductory Network Theory. He* taught an undergraduate course on that subject for seven years, receiving M.LT's Baker Award (Outstanding Teacher of 1963-64). In 1964, he founded the Bose Corporation, which conducts research and development in electronìcs and acoustics, and manufactures high fidelity components. This article is adapted from a presentation given by Professor Bose at a joint meeting of the Acoustical Society of America, the Institute of Electrical a.nd Electronics Engineers (student chapter), and the Audio Engineering Society at M.I.T. last December. The second part of this two-part article will appear in *Technology* Review for July/August.

If the field of sound recording and reproduction didn't have so many experts-many of them not in the discipline of acoustics-I could confine myself to a straight-forward technical presentation of our research. But this field is one in which everybody knows something and almost everybody has some interest and some preconceived ideas. If I just made a technical presentation, some of the results would be so controversial that unless one also knew how they were developed, the mis-match between the preconceptions and the results would be severe. So I'll try to go through the years from '56 to the present. In this manner, I think I can present the developments in the way they occurred to us. The sequence will be evident; and when I arrive at some results that are quite controversial, at least you will see how they came about.
In 1956, I decided to purchase a HiFi. Like many engineers, I purchased it on strictly technical grounds. There was something called "flat frequeney response."

"Flat fiequency response" was by definition a good thing, and one loolced for it. "Uniform phase" was also supposed to be good; to an electrical engineer, fiat frequeney response and uniform phase are, in fact, the criteria for exact reproduction of a signal. "Polar response" was supposed to be round, as I remember, and "distortion" was supposed to be low.
In those days, these data were often printed on a piece of paper on the back of a loudspeaker. Like a good engineer, I normalized them all and picked the optimum speaker. I never consulted a salesman for fear that he had incentives to sell me different things than what the technical world might say were best. I was very confident; as a matter of fact, I didn't listen to the loudspealcer before I bought it, brought it home, and set it up. I even invited some people to hear the debut, for which I had bought some violin recordings.

It was a real shock when the violin started playing, and it was even worse for an ensemble of violins. When they got anywhere up on the E string the sound became sandy, shrill, and screechy. Real instruments never sound that way.

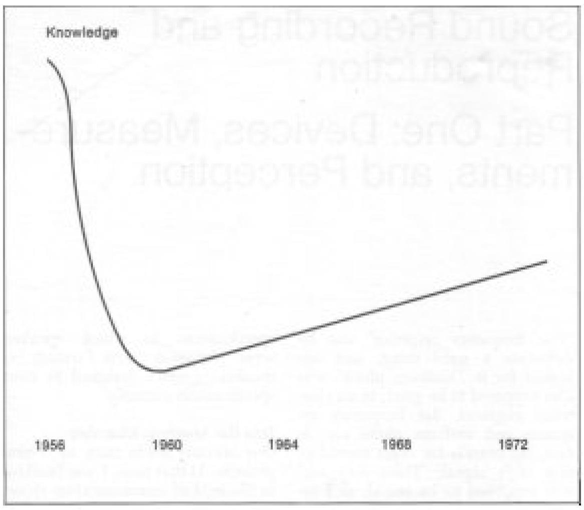
Of course the answer was that I must bave bought a defective unit. So I marched baclc with my speaker, and this time I listened to other ones, more expensive ones. It seemed that the more money you spent, the more screech you purchased. When the salesmen were questioned about this, they would say, "Oh well, people who perform music never like HiFi, you know; they're biased. They don't listen to music in the audience, and this is the sound that HiFi reproduces. . ." Something had to give. Either the specifications to which speakers were designed were wrong, or speakers weren't designed to those specifications correctly.

Into the Anechoic Chamber

Our curiosity led to many night-time projects. At that time, I was teaching in the field of communication theory and I had to do research in the field of acoustics at night. We had good facilities, though: an anechoic chamber-a room with totally sound-absorbent walls-and very good equipment. So we began making measurements on loudspeakers.

One way of depicting what happened thereafter is shown in the form of the curve on the next page, which I think will remain meaningful throughout this presentation. The horizontal axis is time in years; and vertically, we plot knowledge. Our knowledge started off very high in 1956 but by '59 our fund of knowledge was way down; it began turning around in '59; and it's come up a little bit since then. So if at the end of this presentation you know less than you do presently, this is exactly the story of my research from 1956 till now.

Let's start on our way down the curve of knowledge. Let's go into the anechoic chamber and measure a loudspeaker. By the measurement standards in all textbooks and by whatever international standards exist, one goes into an anechoic chamber and places a microphone in front of a loudspeaker, on axis, puts an electrical signal across the speaker's terminals, sweeps that signal across the audio frequency band and plots the amplitude of the microphone signal versus frequency. Supposedly, for a good loudspeaker, the result should be a frequeney response that is fiat.



Knowledge about loudspeaker design: By 1959, it had reached a minimum. "We recognized many fundamental problems, and we understood what was wrong with existing methods . . ."

When we measured loudspeakers, this never happened. The errors weren't two or three dB (factors of abaut 1.5)-they were twenty or twenty-five dB (factors of ten and more). Wc contacted one manufacturer to tell him that we could not measure the same curve that was printed ori the back of his speakers. Without explanation, we were told that we were measuring frequency response incorrectly. After two or three more attempts at communication, with no further responses, we began thinking that perhaps that was the state of industrythey printed one curve but realized another. The thing to do, we decided, was to design a speaker whose frequency response really was fiat. We never questioned the ingrained assumption that flat frequency response is the right thing to have. We wondered how to achieve it rather than if it should be achieved.

We were trying to get it when we began to realize that there was a problem.

What might be wrong with fiat frequency response if we did manage to attain it? The obvious fact is that the environment in which frequency response is measured, the anechoic chamber, in which there are no echoes, is nothing like the environment in which loudspealcers are supposed to perform.

Let's see what happens when a loudspeaker that has fiat frequency response in an anechoíc chamber is placed in a living room. The best way to see what happens is to build one wall behind the speaker in the anechoic chamber as a first step toward converting the chamber into a living room.

Before we put in the wall, let's examine the loudspeaker's radiation pattern. (The illustration at the right). The radius from the origin to a point on the pattern is a measure of the sound pressure at that angle of radiation. It turns out that at low frequencies, where the wavelength of the sound radiated is large (50 Hz has a 23-foot wavelength) compared to a loudspeaker's dimensions, the radiation pattern of any speaker is spherical. It doesn't matter which way the speaker is pointing or what its shape is. The pattern is a circle in a plane; the pressure is equal all around. At high frequencies, where the wavelength is small compared to the speaker dimensions, things don't work this way. In this frequency range, the sound is radiated directionally, as shown by the 15 kHz polar pattern in the illustration.

Flat frequency response in a particular direction requires that at each frequency the speaker must deliver the same sound pressure in that direction for equal-amplitude voltage signals at the speaker terminals. Thus in the illustration the fiat frequency response criterion requires that both radiation patterns must have equal radii along the forward axis.

Now we put a wall behind the speaker in the anechoic chamber. The 15 kHz radiation pattern doesn't change because, as the 15 kHz polar pattern indicated, high frequencies do not radiate behind the speaker. Sound is nothing more than a motion of molecules; if molecules are not moving in any area, an object can be placed there without affecting anything. But low frequencies do go back there. The 50 Hz signal in the illustration reflects from the wall and comes back out again. And at low enough frequencies the reflection is virtually in phase with the original radiation. The wall has therefore doubled the low frequency response as measured in front of the speaker. With just one wall, we have increased the frequency response by 6 dB at the low frequencies. The sound radiated in front no longer meets the criterion of flat frequency response. When we put a floor underneath the speaker, the same efFect occurs again. If we build a corner wall next to the speaker, the low frequency sound pressure doubles yet again. This is, of course, what you have experienced if you have positioned a loudspeaker first in the middle of a room, then against the middle of a wall, then against the wall at the floor, and finally in a corner. Each of those steps will increase the intensity of the bass response of the speaker.

The point of all this is that the textbook criterion toward which we thought we should design-flat frequency response in the anechoic chamber-is incorrect. Even if we or anyone else had succeeded, our wonder speaker would have sounded like a jukebox when brought into a living room. At this point in time (1958) our knowledge was on its way down the curve shown above.

Our first thought was that since a speaker plays in a living room, we'll measure its frequency response there.

However, every room has resonances, caused by sound reinforcing itself through reflections from the walls, floor, and ceiling. The total number of such resonances or "normal modes" in a rectangular room 15 by 20 by 9 feet high is enormousgreater than fifty rnillion over the entire audio band. These normal modes show up as peaks and dips in the frequency response between different points in the room. Each normal mode is like a narrow band filter that either boosts or attenuates the frequency to which it is tuned. Some of the normal modes alter the frequency response by as much as 20 or 30 dB.

We can now see the fundamental problem associated with measuring a loudspeaker in a room. If the frequency response between two points in a room contains a multitude of irregularities, how will we know which irregularities are peculiar to that room and which are inherent in the loudspeaker?

The Space of Sounds

Up to this point wc bave been identifying problems, as wc were doing in 58 and '59. Let's now try to gain some perspective by examining the fundamental nature of problems that involve physical devices, physical measurements, and perception. Consider three abstract spaces (the illustration on page 7,). The first space is a space of devices. By this we mean that each point in the space represents a specific device. For example, point 1 might represent a particular speaker, point 2 a

different speaker, point 3 a turntable, and so on. The second space is a space of measurements. That is, each point in this space represents a specific physical measurement that can be made on a device of space 1. The third space is that of perception. Different points in this space represent perceptibly different performances based upon the same music or speech signals passed through different devices.
Now consider a device such as a loudspeaker. We can make many measurements on it. Frequency response, polar response and distortion are examples of such measurements. In the illustration, we can represent a particular measurement made on a particular device by a line drawn from a point in the first space to a point in the second. Historically, designers bave been preoccupied with creating devices in which certain measurements are optimized. However, they have seldom investigated the relation between the measurements and perception. In other words, the correlations between physical devices and measurements, indicated by the lines joining points in space I to points in space 2, are well established, but there has been relatively little work done to correlate measurements (points in space 2) to the perception of music (points in space 3). Without correlations to perception, the value of measurements made on devices, and the value of designing devices to meet certain measurement criteria, are subject to serious question. To take a simple example: consumers in today's high fidelity market are paying premiums for engineering that tries to optimize measurements that are already below the threshold of audibility. Further lowering of the measurements serves only to improve the performance read from a meter and raise the price. But human hearing cannot detect the improvement in the measurement.

The problem of establishing correlations between points in the measurement space and points in the perception space is deeper yet. There is no reason to believe that useful correlations can be established between measurements and perception when we start with points in the measurement space that were selected only because they represent physical measurements that are convenient to make. Other physical measurements, yet to be discovered, may be more closely related to perception.

Let's look now at another fundamental problem. Consider a space of sounds, in which different points represent sounds that are perceived to be different. Let a point L represent the sound a listener would hear at a certain live performance of music in a concert hall, for example. Then let S,, S_2, S_3, . . . represent sounds the listener hears produced by different music reproduction systems in bis home when he plays a recording that has been made of the performance.
Two salient characteristics of this space are readily established. First, we do not expect to exactly reproduce in the living room the sound a listener would experience at the live performance. There are many factors which preclude this exact reproduction. Prime among them is the living room's small size compared to the concert hall. All sounds produced in a living room bounce around with times between reflections that are very short compared to those of the concert hall. And this difference is easily perceptible.
The second important characteristic of the space of sounds is that we know of no way to establish a measure on it. That is, we know of no way to measure the magnitude of the difference between two points representing two different perceived sounds. In fact, we are a long way from even having a set of measurements that is sufficient to characterize all the different sounds that we can perceive. One can make many measurements on different sound waveforms, but we do not know of any set of measurements that can always predict whether people can detect the difference between these sounds, let alone predict whether a person will judge that sound S_i or S; is closer to L. We conducted many experiments with professional musicians, and all this became quite clear. A good musician can easily identify different sounds-for example, the sounds S_1, S_2, and S_3 produced by different music systems that are playing the same recording. However, assuming that the systems are reasonably good, a strange result will occur when you ask the musician which system is the most accurate-which of the points S_l, S_2, and S_3 is closest to L in the space of sounds. One musician in our experiments commented: "I can distinguish between the systems easily, but I cannot say which is more like the live performance. It's as if you asked me whether a peach or a grapefruit is closer to a lemon. The peach has more the size of the lemon and the grapefruit has more the appearance of the lemon." If we do not have a measure on fhe space of sounds and we cannot reach the perfection of recreating the concert hall performance in the home, then we must ask: How close can we come to our goal? Without knowing how to measure the distance to exact reproduction, we could work forever without realizing that we might already be as close as we can get to this goal.

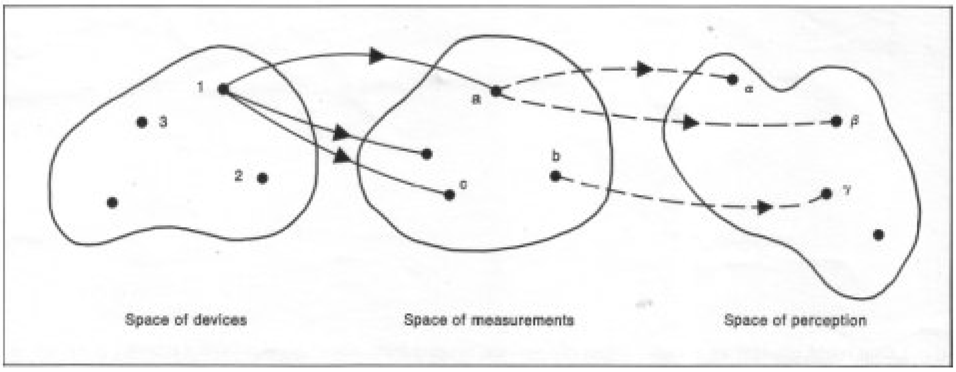


fig.: A way to visualize the fundamental nature of problems involving devices, measurements, and perception. Lines connecting points in the space of devices to points in the space of measurements represent specìfic measurements made on specìfic devices. Points in the space of perceptions represent perceptibly different processing of the same signal by different devices. Measurements do not always correlate uniquely with perception. One common measurement (symbolized above by a) is made by presenting a sine wave at some frequency f_o to a device, which, in processing the signal, introduces distortion. A notch filter removes the f_o component from the output. The waveform that remains is the distortion. The ratio of its RMS amplitude to the RMS amplitude of the input constitutes a measurement of "harmonic distortion." But the distortion can be divided in perceptibly different ways among the harmonics of f_o; the illustration immediately above shows two possible distributions of harmonics yielding the same measurement of harmonic distortion. Measurements have often been selected because they were convenient. There may be other measurements, as yet unknown to us, with signifícant relations to perception.

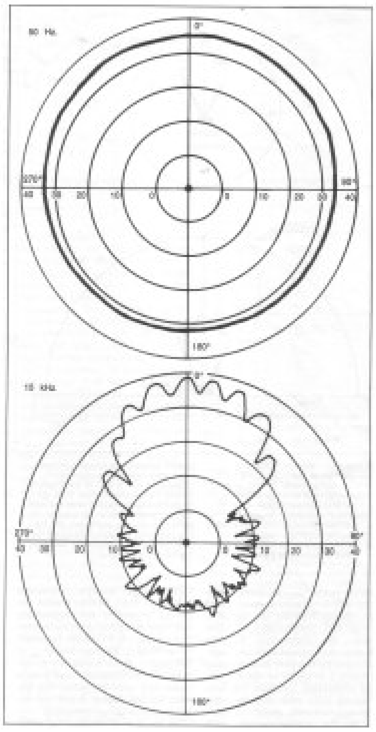


fig: The radiatìon patterns, at a low frequency (top graph) and a high frequency (bottom graph), of a loudspeaker in an anechoic ahamber. Six dB difference between two measurements corresponds to a doubling of sound pressure. At 15 kHz, the pressure radiated forward (at 0°) averages about 25 times greater than pressure radiated 60° or more off axis. At 50 Hz, however, pressure is much closer to equal in ali directions, off by only about 30 per cent even behind the speaker.

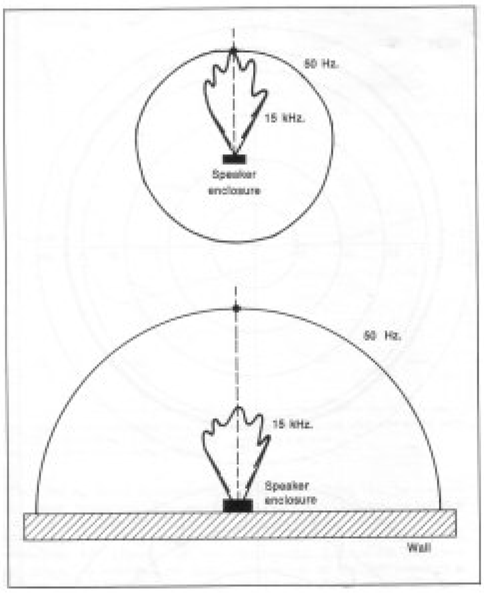


fig: Flat frequency response in the anechoic chamber is lost in the living room: The top drawíng shows stylized radiation patterns-one at a low frequency, the other at a high frequency-for a loudspeaker in an anechoic chamber. The patterns coincide on the forward axis; thus the speaker's frequency responses are equal at those frequencies and in that direction. As a first step toward simulating a listening environment more typical than an anechoic chamber, a wall is built in the chamber behind the speaker (bottom drawing). Since the high frequency does not radiate appreciably behind the speaker, its pattern is virtually unaffected. But the low-frequency radíatio-n reflects off the wall, doubling the frequency response at the low-frequency end.

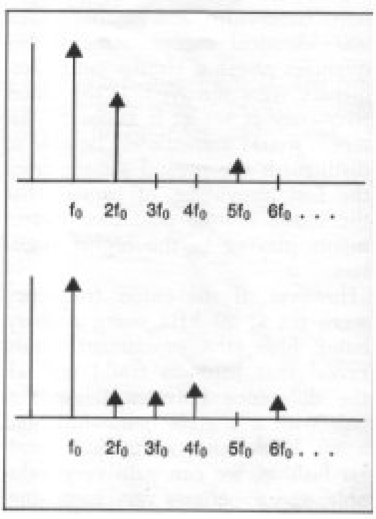
Difference Experiments

By 1959 we were at a low point on our curve of knowledge vs. time. We recognized many fundamental problems and we understood what was wrong with existing methods of speaker design, but we had no constructive approaches to offer. In fact, we were about to open our parachutes and abort the research program.

Coming from mathematical disciplines into a discipline like this where everything seemed soft ... it just didn't appear that we could get a handle on it. We didn't want to muddle through by asking people their opinions and finally creating a system that one particular person or a group of people happened to like. We wondered whether we could do
anything scientific. We knew we couldn't find out which loudspeaker was best, but could we determine ingredients of speakers that would contribute to more accurate reproduction of music?

Pursuit of that approach proved very fruitful. If we properly limit the questions we ask, we can indeed gain much useful information about various design parameters and their correlation to perception. In fact, as we shall see, we can even determine the limits for optimization of different design parameters beyond which there is no audibly perceptible change in performance. The key that enabled us to obtain this valuable knowledge is a basic concept in psychoacoustic testing which we shall call a "difference experiment." In these experiments, listeners are given a sequence of two musical signals differing at most in only the one parameter under investigation. The listeners are asked to indicate only whether they can detect any difference between the first and second signals of the sequence. By arranging an ensemble of these sequences in which the first signal is common to all and the second is sometimes identical to the first and otherwise is identical except that the parameter under study has been altered, one can readily determine the range of audibility of the given parameter.

Let's take a very simple case and see what can be learned. If all the knowledge we nów have about the hearing process were erased, one might arbitraríly decide to design a HiFi system with a frequency response out to a megahertz. How could one learn that was not necessary? (It could get pretty expensive.) One could conduct an experiment in which listeners are given one music signal with a one-megahertz bandwidth, and another which was identical except that all frequencies above a chosen cutoff frequency were removed. If the cutof frequency is set at 5 kilohertz, listeners would immediately be able to distinguish the second sample from the first, providing, of course, that the music sample contained instruments playing in the higher registers. However, if the cutoff frequency were set at 20 kHz using a sharp cutof filter, the experiment would reveal that listeners could not tell the difference between music signals with a 1 MHz bandwidth and a 20 kHz bandwidth. In a similar fashion, we can gain very valuable, and sometimes very surprising, information about many other parameters. In all these experiments, great care must be talcen to ensure that the only parameter that changes in the sample set is the one under study. This may be obvious, but historically it has been the cause of many erroneous conclusions- everal parameters change but t Me experimenter makes conclusions about just one of them.



The Ideal Pulsating Sphere

The concept of a "difference experiment" can be used in much more sophisticated situations to yield very significant information . about speaker design. One such experiment occupied our research efforts from 1960 to 1964.

There has always been a great deal of mystery and controversy associated with the choice of methods of converting electrical energy into acoustical energy. Articles have appeared claiming benefits for various

types of speakers, ranging from electromagnetic through electrostatic to ionic units. It has been common belief, though, that none of the methods could produce music in a room as purely as could the unrealizable "ideal pulsating sphere" cited in acoustics studies. This ideal sphere is, by definition, a theoretically perfect moving surface that has no resonances or distortion and that launches a sound wave that is an exact replica of the electrical waveform that is its source.

However, experiments we had conducted from 1956 to 1960 indicated that this common belief might be in error. We thought it might be possible to construct on a spherical surface an array of small full-range speakers that would (with appropriate contouring of fiequency response by means of electronics) reproduce musie which would be subjectively indistinguishable from that which would be reproduced by the ideal pulsating sphere. BLIt how could we prove or disprove this since the ideal sphere is something that -cannot be constructed? This was the subject of our work for four years, and a difference experiment was instrumental in resolving the issue.

Of course, if we had access to an ideal sphere we could perform our experiment by making recordings in a room, first of the ideal pulsating sphere's performance and then of the real spherical array of speakers playing the same musical selections. We could then have the recordings played over high-quality headphones and make the standard "A-B" comparison tests to see if listeners could distinguish between the two recordings. The procedure would be relatively straightforward-but we obviously do not have access to an

ideal pulsating sphere.

However, with the aid of a highspeed digital computer-the TX-2, designed by the M.I.T. Lincoln Laboratory-we were able to obtain recordings of the sound that an ideal pulsating sphere would produce in a given room. These recordings were then used in the comparison tests with our array of speakers.

A few words about how the computer was used te produce the recordings of the ideal sphere: Since the propagation of sound in a room-even at the highest listening levels-meets certain conditions (mathematically speaking, meets the requirements for linearity), we could use the techniques of linear system theory to obtain our desired recordings. In particular, if we knew only the response of a linear system to a narrow pulse, then we could calculate the response of that system to any input. The mathematical operation by which the response is calculated is called. convolution.

Thus we needed only to construct an acoustical source which could produce the same pulse as would be produced by the ideal sphere. The recording of this pulse in a room would then enable us to calculate the response of an ideal sphere in that room to any music or speech signal.

After trying many sound sources, we finally determined that an electrical sparle discharge would meet all the conditions required for our experiment. Thus, a spark was set off in a room in a position where the ideal sphere would have been placed. The acoustical result of this spark was recorded through a microphone placed about ten feet back in the room. This recording was sampled 30,000 times a second, converted to a digital signal, and stored in the TX-2 computer, where it was convolved with music and speech sources that were later fed to the computer. The computer thus became a sort of HiFi-it produced audio tapes identical to what the microphone in the room would have recorded if the music and speech samples had been played in that room through an ideal pulsating sphere. In this way, we obtained recordings of a range of music and speech as played through the ideal sphere-without ever having the sphere.

While this experiment is simple in concept, the experimental difficulties were enormous. Moreover, the experiment required extension in the computer programming knowledge existing at that time. One computer programmer had originally estimated for us that it would take hours of computer time to produce one second of music. Professor Thomas Stockham developed a program which was dependent upon the data in a way that shortened the computation time to seven minutes per second of music. Without his program, the experiment could not have been conducted.

The results of this four-year difference experiment were presented to a joint meeting of the Institute of Electrical and Electronics Engineers and the Audio Engineering Society at M.I.T. in November, 1964. The conclusion was that no one could tell the difEerence between the performance of the array of small fullrange speakers on the spherical surface and the computer simulation of the performance of the ideal pulsating sphere.

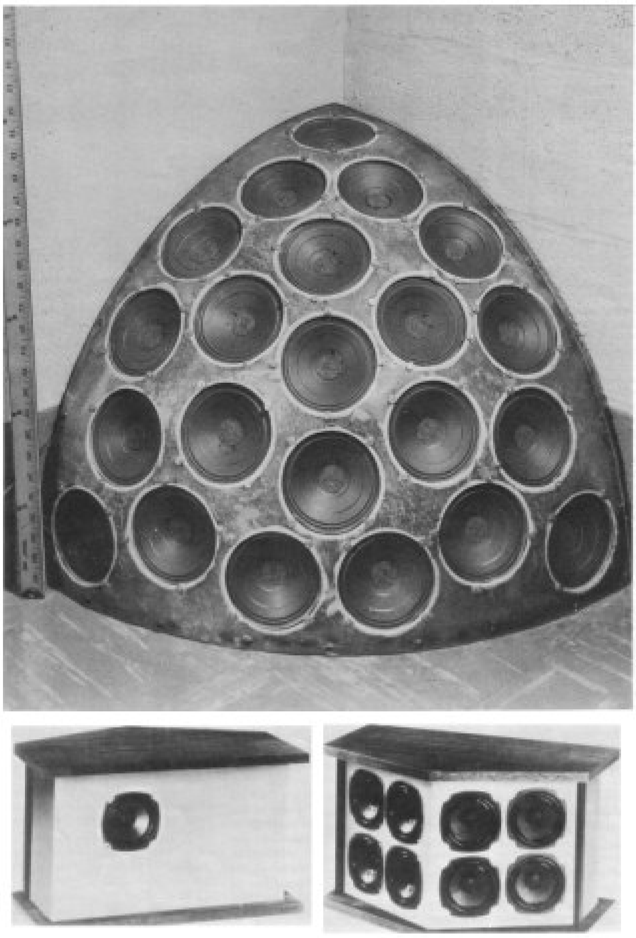
Two very basic results were established by the experiment:

Ci The experiment proved that, as far as quality of reproduction is concerned, we need not look beyond a multiplicity of full-range cone speakers, since they are capable of reproducing music and speech signals that are subjectively indistinguishable from those reproduced by the ideal pulsating sphere. Thus the search for other methods and materials for more accurate music reproduction could be ended. While it might be possible for other transducers to do as well as the multiplicity of cones, they could not do audibly better.

E Since the ideal pulsating sphere has, by definition, no audible distortion, and no audible irregularities in frequency response, transient response, or other responses, the experiment proved that the measurable irregularities of the array of full-range cone speakers did not contribute any audible coloration to the reproduced signal (since that signal was subjectively indistinguishable from the performance of the ideal sphere) .

We were quite pleased with the knowledge wc had gained from this experiment. But we were shocked by a totally unexpected result: The spherical array of speakers still exhibited objectionable shrillness on ensembles of violins and wind instruments playing in the higher registers. We had always thought that shrillness like that I had heard when I purchased a loudspeaker in 1956 was associated with either distortion or frequency response inegularities. Now, for the first time, we realized that it was not caused by any of the parameters we normally measure in loudspeaker design, because the spherical array sounded identical to the ideal pulsating sphere-which, of course, had no irregularities in any of the normally measured parameters. There had to be other di mensions to speaker design and measurements that we had not yet discovered.

(First of a two-part article. The conclusion will appear in Technology Review f *or July/August.)*



An ideal pulsating sphere is simulated by 4he eighth of a sphere shown at upper left. The eighth sphere in a corner of a room is equivalent to a full sphere floating in empty space, since ideai walls are to sound as ideai mirrors would be to light. Below left, front and rear views of a loudspeaker designed to radiate mos2 of its sound backward, to reflect off the wall behind the speaker.

Sound Recording and Reproduction
Part Two: Spatial and Temporal Dimensions

This is the second in a series of two articles on sound recording and reproduction by Professor Bose. Part One appeared in *Technology* **Review for June, 1973, pages 19-25. Some of the research reported in this series was made possibte in part by support extended the Massachusetts Institute** of **Technology, Research Laboratory of Electronics, by the Joint Services Electronics Programs under Contract** No. DA **28-043-AMC02536(E).**

In 1964, we had created a speaker whose sound was subjectively índistinguishable from the sound of an ideal pulsating sphere, which we had simulated with the aid of a digital computer. But when music was played through this speaker, it exhibited many of the shrill, harsh sound characteristics that we had heard in conventional loudspeakers. It might at first appear that after eight years of research we were back where we began. That is, we had created a speaker with undesirable characteristics similar to those of speakers we had studied at the outset of our research.

In fact, we were at this time far ahead of our position in 1956, because we were now able to rule out certain factors as possible causes of the undesirable sound. The idea] pulsating sphere has, by definition, flat frequency response, perfect transient response, no distortion of any type, and omni-directional radiation. Since our creation, a loudspeaker consisting of an array of 22 radiators on a sphere, sounded identieal to the simulated ideal sphere, we now knew that the undesirable sounds emerging from our speaker were not caused by frequency response, transient response, distortion, or by any deviations from omni-directional radiation that the speaker may have had. This realization was indeed surprising, since these four characteristics were thought to be the determinants of the sound quality of a loudspeaker. Yet we had now encountered a situation in which these characteristics were literally beyond question, but the undesirable nature of the sound remained. This marked the first time that we could conclusively state that attempting to improve the frequency response, transient response, distortion, or omni-directionality of radiation of our speaker design would offer no further audible benefits.

There was only one conclusion that could be drawn from the experiments that were completed in 1964; There must be other parameters important to hearing that had not yet been considered in speaker design. In 1965, we set out in search of these parameters.

Binaural Recording

We began by studying the method of sound recording and reproduction that enjoyed the reputation of being the most accurate: the so-called binaural method. Binaural recordings are made with two microphones placed in the ears of a dummy head which is positioned in an audience at a live performance. Reproduction of the performance is accomplished by playing back the signals recorded from the microphones in the dummy's left and right ears through headphones on a listener's left and right ears, respectively. In this manner, one attempts to duplicate at the listener's ears the acoustical signals that were present at the ears of the dummy recording head at the live performance.

If you have ever heard good binaural recordings, you know that they are superior in many aspects to the sound you have heard from the best loudspeaker systems. The principle limitation-and it is an important one-of binaural recording and reproduction resides in the fact that signals at the ears of the listener made by binaural recordings do not change with motions of the head as signals do in the live performance. The failure of binaural reproduction to produce correctly changing signals with head motions is responsible for the often-observed sensation that the sound source is within the head of the listener rather than external to it. Our first major experiments with binaural recording were performed at the Tanglewood Music Festival with the cooperation of the Boston Symphony Orchestra. The dummy recording head was seated fifth row center, and two track recordings were made of the signals from its left and right ear microphones. On playing our recordings, it was immediately apparent that the reproduced binaural signals did not exhibit the shrill, harsh sounds that were characteristic of the loudspeakers.

The most significant result of our experiment occurred when we added the left-ear and right-ear signals to make a monaural signal that we applied identically to each ear of the listener. When this monaural signal was played over the headphones, the shrill sounds were again apparent. Many musicians and conductors of orchestras went through this binaural-monaural listening experience for us, and the results were always the same. In fact, when we switched the binaural signal into a monaural signal while one of the guest conductors at Tanglewood had our headphones on, he complained: "What did you do? All the evils of my HiFi have returned."

The dummy head used by the author and colleagues to make binaural recordings at the Boston Symphony Orchestra's Tanglewood Music Festival. Microphones in the dummy's left and right
Il ears" record the summation of sound pressures that would be present at the ears of a human concert-goer.

It is important to note that in this A-B experiment between binaural and monaural sound, the conventional parameters such as frequency response, transient response, and distortion of the recording and playback apparatus were unchanged between the binaural and monaural signals, since the monaural signal was formed by simply adding the two binaural channels together. This is exactly the type of experiment we wanted-an experiment in which the basic parameters that we had previously studied remained fixed and new parameters changed. Without yet knowing why it worked, wc were able for the first time to suppress the harsh, shrill sounds without also losing desirable properties of the msuic. We then looked at the binaural system and interpreted the results of our experiment.

In a live performance, the sound waves radiated by musical instruments reflect from all surfaces of a concert hall and arrive at a listener's ears from all directions. Consider a sound wave arriving at the head from any given direction. Except in the case of sound sources exactly in front of or behind a listener, this wave arrives at one ear before it reaches the other, since the path lengths to the ears are unequal. In addition, depending upon the frequency range of the arriving wave, the spectrum of the signal that results from this wave will be different at the left ear and the right ear. High frequencies, for example, will be attenuated as they diffract around the head. Thus we see that, in general, there are time and spectral differences between the signals at the two ears for any given direction of the arriving sound waves.

In making a binaural recording, we measure the pressure at the plane of the pinna of each ear caused by the summation of all arriving waves, and attempt to reproduce this pressure at the corresponding ear of the listener through the use of earphones. Now imagine that we have added the left and the right channels together, then fed this signal to both headphones. The listener receives the same signal at each ear. This could be achieved without headphones by simply beaming the sound from a nearby loudspeaker directly toward the face of a listener. Our experiment had shown that the monaural signals produced a sensation of shrill, screechy sounds typical of conventional loudspeakers, and that when the headphone signals were changed to the binaural signals, these undesirable effects disappeared. This strongly suggested that the undesirable effects produced by loudspeakers might be related to the distribution of angles at which the sound they produced arrived at the listener. In fact, it suggested that beaming sound waves at a listener might be the cause of the harsh, shrill and unnatural effects.

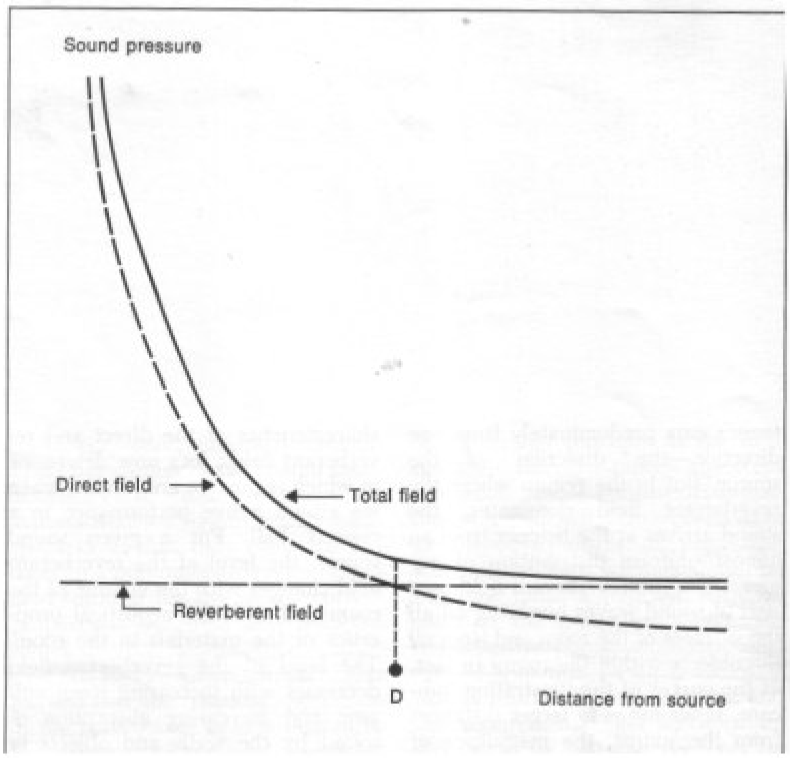
Motivated by the result of this binaural experiment, we shifted the focus of our research to the spatial aspects of the sound waves incident upon a listener. We conducted a series of experiments designed to determine the effect upon perception of varying the distribution of the angles of incidence of sound waves on a subject. One of our first experiments of this type involved placing five similar direct-radiating loudspeakers around a subject, and conducting the experiment so that the subject would first experience one loudspeaker directly in front of him, and then experience the sound from all five loudspeakers adjusted to create the same frequency spectrum at the position of the listener as the single loudspeaker had created. (This experiment must be carried out in a nearly anechoic environment.) We found that the undesirable harshness and shrillness observable from.the single loudspeaker was significantly reduced when all the loudspeakers were simultaneously playing. This experiment and others indicated the importance of bringing the sound waves to the listener from a wide distribution of angles, which of course, was just the opposite of what had been practiced in loudspeaker design.

Direct and Reverberant Fields

The results of a number of spatíal experiments encouraged us to study the characteristics of the sound waves incident upon the listener in concert halls. We soon discovered that a wealth of information already existed on this subject, resulting from decades of research in architectural acoustics. Although that research was directed primarily toward the design of large halls for live musical performances, it is not surprising that an understanding of the acoustics in a live performance can yield valuable information for the design of home music systems.

It is convenient to divide the sound incident on a listener in a concert hall into two categories: direct sound and reverberant sound. Direct sound is defined to be that component of sound that arrives at the listener's ears direetly from the source-a musical instrument, for example-without any reflections: the sound thaí travels the straight path between the instrument and the listener. the reverberant sound is defined to be the sound that reaches the listener after one or more reflections from either the surfaces of the concert hall or objects within the hall. Clearly, by these definitions the direct sound and the reverberant sound encompass the totality of sound arriving at a listener's ears from the musical instrument.

Now consider, for convenience of analysis, the fields produced by an omni-directional source of sound suspended over the stage of a concert hall. the sound pressure of the direct field radiated from the source to the audience varies inversely with the distance between the source and a listener, as shown in the illustra



The total sound field and its direct and reverberant components. Direct sound is that which travels the line from the source to the listener, without reflections. Reverberant sound is that which arrives at the listener after one or more reflections. At a certain distance D from the source, which depends on the nature of the room, the contrlbutlons of sound pressure from the two fields are equal. (But their sum is not double the pressure of either. Because the fields are un-correlated, they are combined by taking the square root of the sum of their squares.)

tion on this page. This is the same situation that would prevail if the source and listener were located outside, and there were no way for sound to arrive at a listener via reflections. The sound pressure of the reverberant field, however, behaves very differently. Except for local maxima and minima at specific frequencies associated with the norma] modes (resonances) of the hall, the sound pressure in the reverberant field is essentially uniform, independent of position in the room, as shown in the illustration.

The two curves reveal some interesting aspects of the total sound field and its direct and reverberant components. Notice that at a certain distance from the source (marlced D in the illustration), the sound pressure of the direct field is equa] to that of the reverberant field. At distances from the source shorter than D, the sound pressure of the direct field dominates and the sound pressure increases rapidly as we approach the source. At distances from

the source greater than D, the reverberant field's sound pressure dominates and, in fact, the total sound pressure becomes virtually independent of distance from the source. You have experienced in your home a consequence of the shape of this curve of total sound pressure versus distance if you have ever listened to a radio, first with your ear very close to the loudspeaker and then while wallcing away from the speaker across the room. When you are very close to the loudspeaker-within a foot or so-it is quite loud; that loudness decreases as you walk away, but depending upon the size of your room, after you are three to six feet away from the speaker, the volume level remains about constant as you move across the room.

The spatial natures of the sound fields in the regions closer to or farther away from the source than D are very different. In the region where the direct field dominates, the sound obviously arrives at the listener's ears predominately from one direction-the direction of the source. But in the region where the reverberant field dominates, the sound arrives at the listener from an almost uniform distribution of angles, since the reverberant field consists of sound waves bouncing ofF all the surfaces of the room and from al] the objects within the room. In fact, as the curves of the illustration indicate, as we move to larger distances from the source, the magnitude of the direct field sound pressure becomes negligibly small compared to that of the reverberant field. At these larger distances, the sound pressure of the waves incident upon a listener would be at essentially the same level for all angles-including the angle corresponding to the direction from the source, since the magnitude of the direct field is so small compared to that of the reverberant field.

Another important difference between the fields in the two regions is in their frequency spectra. The frequency spectrum close to a source-that is, in the region where the direct field dominates-is that which the source radiates along the axis from it to the measuring point. In the reverberant field, however, the frequency spectrum is related to the total acoustic power that the source radiates in all directions, since it is that total power which is the source of the reverberant field. ,)~(Clearly, the absorptiorr characteristics of the room enter into the spectrum of the reverberant field, but the point we wish to make here is that the total power radiated from the source, rather than the radiation along any particular axis, is what creates the spectrum of the reverberant field).

Having examined some of the basic characteristics of the direct and reverberant fields, let's now determine in which region wc are located when wc attend a live performance in a concert hall. For a given sound source, the level of the reverberant field changes with the volume of the room and with the acoustical properties of the materials in the room. The level of the reverberant field decreases with increasing room volume and inereasing absorplion of sound by the walls and objects in the room; and the distance from the source at which the sound pressures of the direct and reverberant fields are equal becomes larger. For omnidirectional sources, this distance varies from a few feet in small rooms to 19 feet in Boston Symphony Hall. For directional sources, the distances are greater, depending upon the degree of directionality of the source. What is clear is that virtually all of the audience in a concert hall is seated in the region where the reverberant field is dominant. Therefore the spatial and spectral aspects of the sound incident upon the listener in the audience are those we have described for that region. The physical characteristics of the sound field might at first seem to contradict human perceptual abilities. When we consider the result that the reverberant field is dominant
in the audience and that it arrives with virtually equal intensity from all directions, we are tempted to con clude that this situation would preclude our ability to locate the direc tion of musical instruments, or even the direction of the stage upon which the instruments are playing. Of course, this is not the case. While an instrument plays, it is continually emitting new acoustical signals. Each of these signals reaches our ears first by the shortest path from the instrument to us and then, tens of milliseconds later, arrives repeatedly via reflections as part of the reverberant field. It is well established that we make use of the first arriving wave-that is, the direct field-for detecting direction and that we can localize the source via this first wave even though its sound pressure is far below that of the reverberant field. This localization ability is similar to that involved in the so-called

cocktail party ef£ect: in the midst of a gathering of people, we can, if we so choose, localize and focus on one conversation in the presence of an ambient sound level that is much higher.

The reverberant field, while contributing nothing to our ability to localize, plays a very important role in our perception of the timbre of music. Since the reverberant field is dominant in the concert hall, it determines the ratios of harmonics to fundamentals, and therefore the timbre of musical instruments. (Angles of incidence and temporal effects also strongly influence our perception of the timbre of instruments, but these, too, are associated with the reverberant rather than the direct field.)

Loudspeaker Design

Our original motivation to study the sound field in a concert hall had come from binaural recording and other experiments that indicated that many of the harsh and shrill sounds encountered in loudspeaker music reproduction may be due to the fact that loudspeakers have been designed to beam the sound directly at the listener, as opposed to delivering it to the listener from a wide distribution of incident angles. Our study of the concert hall had revealed that sound does in fact arrive at the listener's head with virtually uniform distribution of angles. Thus, our object should be to design a speaker that radiates only a small portion of its energy directly toward the listener and grazes the larger portion of its radiation off the walls of tloe listening room to create spatial patterns in that room comparable to the spatial patterns one receives in the audience of a live performance. Experimenting with this concept, we found that excellent results were obtained by employing about ten per cent direct radiation toward the listener and about ninety per cent of the radiation at 30° angles toward the wall behind the loudspeaker. In terms of a practical design, we accomplished this by a loudspeaker (shown on page 10) with a single driver mounted on a front panel and four similar drivers mounted on each of two rear panels that array those drivers at a 30° angle to the rear wall, which now functions like the stage wall behind the instruments in a live performance. Since we are much closer to a loudspeaker than we are to the instruments in a live performance, our loudspeaker design called for a large ratio of reflected to direct sound. The small amount of direct sound is al] that is required for localization.
To summarize what we had learned from our research:

- *Multiplcity of Full Range Speakers.* From the spark experiments described in the first part of this article (which appeared in *Technology Review* for June), we learned that it is possible to produce music without audible coloration from distortion, resonances or transient response irregularities by the use of a multiplicíty of full range speakers.

- *Direct and Reflected Sound.* From our studies of the spatial characteristics of sound fields, wc learned that wc should design loudspeakers so that they place the listener in a predominantly reverberant field through the use of the correct proportions of direct and reflected sound.

- *Flat Power Criterion.* We also learned that the frequency spectrum of the reverberant field is related to the spectrum of the total power radiated by the source or loudspeaker. Therefore, if we are to achieve the same balance of frequencies in our listening room as we experienced in the live performance, the loudspeaker should be designed to a flat power criterion rather than the conventional flat-frequency-responseon-an-axis criterion.

- *Equalization.* Loudspeakers do not inherently meet the criterion of fiat power response, since it has been traditional to design them to approach flat frequency response.

However, with proper equalization, we can adjust the frequency balance of the electrical signals entering the loudspeaker so that the sound waves that emerge will satisfy the flat power criterion. This can be accomplished with the aid of an electronic frequency equalization network.
Having incorporated our research findings into the design of a loudspeaker system, we were again confronted with the basic problem we discussed in Part One: there exists no objective measure for the space of sounds, and therefore there is no objective way of telling whether the performance of one design is superior to that of another-let alone whether a design is optimum. The only thing we could state was that, based upon our research in physical acoustics and psychoacoustics, each of the design concepts in the new speaker should represent steps toward the more realistic reproduction of music. We did not even have an objective means of predicting whether the steps we took

would be small or quite significant with respect to the perceived sound. Ultimately the public would become the judge of this. In 1968, manufacture of this speaker was started, and the acceptance which followed indicated to us that the concepts involved produced changes that people regard as significant improvements.

Looking Ahead: Sound Reproduction ...

Where do we go from here? Have we optimized all the parameters for music reproduction in the home, or can we expect to do a significantly better job in the future? We might feel compelled to say that of course we can do better in the future, because such is the nature of progress. However, with respect tfl specific tasks, research can and does bring the state of affairs to a point where further work toward that specific task results in diminishing returns. For example, it is safe to say that electronics technology has brought the state of amplifier design to the point where well-designed amplifiers introduce no audible coloration into the signals they amplify. This situation will not be improved upon in the future by any technique. But progress may well bring amplifiers of smaller dimensions, lighter weight, less power consumption, and possibly less cost.

Our answer to the question, "Can we expect better home music reproduction in the future?" is a resounding yes. We thinlc that the next five to ten years can bring dramatically superior music reproduction into the home, but it may come through different avenues than one might at first expect.

In looking at what has been accomplished as well as what remains, it is convenient as well as accurate to organize the problems into three groups:

- *Speetral Problems,* under whieh we shall include, by definition, the topics of frequency response, distortion, and transient response.

- *Spatial Problems:* the distribution of angles of incidence of sound upon the listener. This is not to be confused with the localization of sound by the listener, which can be affected by the balance of stereo channels in the recording process. Here we are simply involved in geometrical considerations relating the loudspeakers, the room, and the listener.

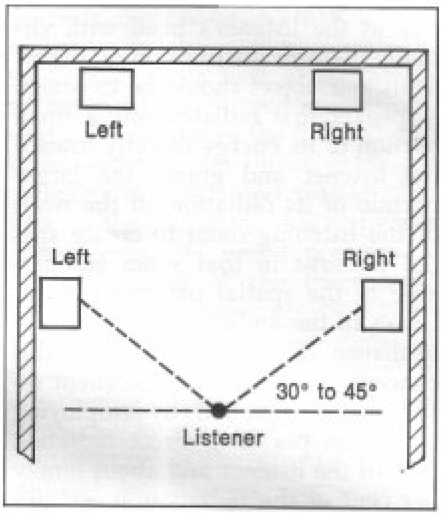
- *Temporal Problems:* those eoncerning the environmental characteristics that enable us to distinguish the acoustics of a large hall from those of a small room. We make this distinction using primarily characteristics of the music signals that are best described in the time domain, as we shall see.

We will first consider the problems of reproduction of sound. the results of the experiment that simulated the ideal pulsating sphere, described in Part One, are sufficient to assure us that the spectral problems of loudspeaker design are well in band. That experiment showed us that through the techniques of multiplicity of full range speakers and equalization we can reduce the problems in the spectral category to a point below the threshold of audibility. In practical applications, one can expect small continued improvement due to minimizing the tolerances in production units, but at the research level the spectral problems can be considered solved.

There is room for significant improvement in the spatial characteristics of home music systems. After all, we are quite lucky to achieve any decent results when we attempt to reproduce the sound of a 70-piece orchestra in a large concert hall through two loudspeakers in our living room. Yet we do not expect that we can make large improvements over the spatial characteristics of a loudspeaker of the type developed in our research. The kind of spatial improvement needed might be produced by an enormous radiating surface which covered the front wall of a room and overlapped around the side walls to produce the sensation of sound from a 50-foot stage. This impractical transducer can be approximated (the illustration above) with satisfactory results by a second pair of speakers against the side walls of the listening room positioned 30 to 45 degrees ahead of the listener. Thus we expect the major improvements in the spatial aspects of music reproduction to come from the use of additional speakers rather than from improved design of the individual speakers.

The temporal problems offer the greatest opportunity for improvement of music reproduction. As we shall see, the temporal parameters are not under the control of the speaker designer. Rather, they can be affected through additional channels of information anMd signal processing yet to be developed.

To understand the temporal parameters, consider the following experiment. Imagine that you are blindfolded and taken into a concert hall and into a much smaller room. In each case, you are asked to speak or



A way to simulate the sensation of sound emanating from a large stage, without having a sound-radiating surface that wraps around the living room walls. Each stereo channel drives two speakers.

to listen to the plaving of a live instrument. You wouÌd easily be able to tell which was the concert hall and which was the small room. It is sometimes said that the distinction is made on the basis of the longer reverberation time of the concert hall. While you would certainly perceive this, it is not the dominant factor that enables you to judge the size of a room. Tiled shower rooms with dimensions of only 15 to 20 feet on a síde will often have reverberation times significantly greater than the reverberation times of concert halls, yet you will be aware that you are in a small room when you stand blindfolded in the shower room. It turns out that the times *between* reflections of sound are more significant parameters for sensing the size of a room. And the mean time between reflections in a large concert hall is an order of magnitude greater than that in an average living room.

This points out a significant problem for home music systems attempting to appmximate a concert hall performance. All the sounds emitted from the speakers in the listening room bounce around the room with a mean time between reflections typically as small as 7 to 12 milliseconds, depending upon the size of the room. Thus, we cannot expect any pair of stereo speakers located in a living room to provide the temporal sensations that are so important in a live performance.
To a small extent, some sensation of reverberation can be introduced into recordings. However, this technique is very limited in its effect, for when the reverberated sound is emitted from the same location as the original signal, and when it bounces around a small room together with that signal, the result tends to be loss of definition and otherwise unpleasant effects long before the amount of added reverberation in the recording becomes comparable to what one would experience in the live performance. To successfully give the ímpression of a large hall to a listener in a relatively small room, it seems that it will be necessary to employ additional speakers located toward the rear of the listening room and to drive those speakers with signals that will give the listener some of the sensations that he gets from the sound waves that bounce ofF the side and rear walls of the concert hall. The signals that are fed to these additional speakers will be different from those fed to the two front stereo channels. Each different signal must be amplified by an additional channel of the music reproduction system.
It does not follow that the proper signals to apply to these additional channels would be recordings made
further back in the concert hall. Such recordings would experience the reverberation of the hall as well as that of the listening room to yield a very undesirable result.
Fundamentally, it appears that all the information necessary to derive the signals for the additional channels resides in current stereo recordings. To achieve our desired result, we want to record signals close to the orchestra, as done in many current stereo recording techniques, and process these signals in such a way that when they are played back over additional channels in a small

room, they will produce some of the sensation that one would experience in the larger concert hall. It is not yet certain that this can be achieved, but based upon some research that we have already conducted, the prognosis is good. At this point in time, the signal processing would require more equipment in the home than is presently feasible. However, with the rapid development of integrated circuits, it would not be surprising if signal processors for deriving additional channels were available in a few years in the size and price range of current receivers. One system of adding additional channels for home music reproduction is known commercially as quadraphonic sound. While the general principle of additional channels has excellent prospects, the present practices leave much to be desired. There are various systems using socalled matrixing methods available today, which attempt to combine four channels into two for broadcast or recording and recover them at the receiver, and which also synthesize four channels from the two channels of an ordinary stereo broadcast or recording. The problem of compressing four channels of information
into two without increasing the bandwidth or dynamic range of the existing channels is analogous to the problems faced 25 years ago when the additional information required for color television was successfully included in the bandwidth previously used for only black-and-white. The difFerence is that the intensity and the competence of the research applied to solving the problems in color television was far greater than that which has been applied in the audio industry to the problem of quadraphonic sound. It is safe to say that none of the existing matrix methods are well founded from a technical point of view, and that their performance is consistent with the level of their technical development. Quadraphonics as presently practiced is a striking example of the danger in allowing marketing departments to establish standards of technology based upon the very first efforts of the engineers. Should the pressures of industry result in the Federal Communications Commission accepting one of the present matrixing methods, the great potential benefits that reside in multiplechannel music systems might never be realized.

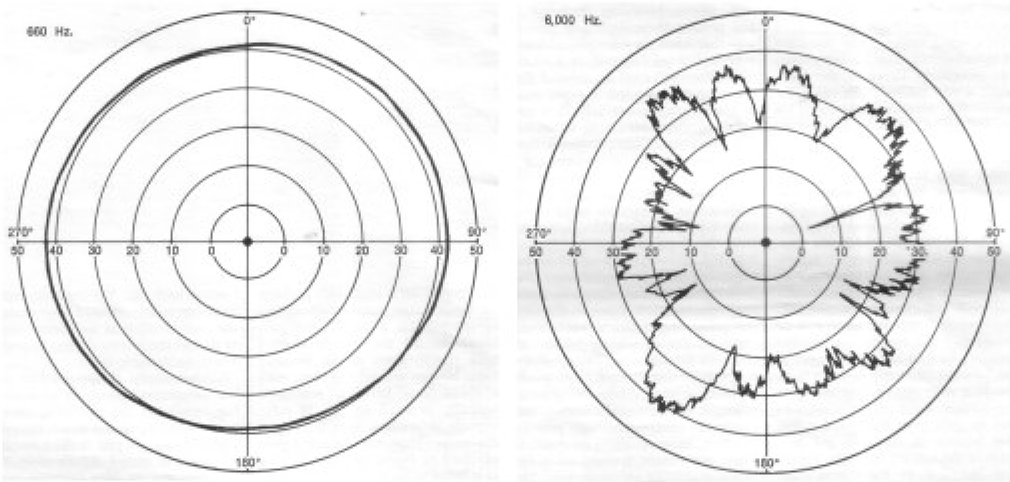
On the brighter side, I suppose that one can rationalize that the existing systems of quadraphonics have a role if whatever they produce is sufficient in the judgment of the consumer to justify a purchase. In this connection, wc must always keep an open mind with respect to the objectives of any medium of entertainment. While most of our work has been devoted to more accurately reproducing the sensations of a live musical performance, there are certainly valid pursuits which simply create interesting effects with the hope that artists will learn to use these new systems much as they would learn to use new instruments.

... And Recording

In looking at the future of home rnusic systems we must also consider the techniques of recording. The problems fall into two categories: equipment limitations and limitations of the acoustic field in the recording environment.

With respect to equipment limitations, the most salient problem is the dynamic range problem, which manifests itself as noise level that is obvious during quiet passages of music. The smallest amount of audible hiss on a recording, for example, detracts markedly from the realism of the reproduction. In research situations, which are fewer tape copies removed from the original performance than are commercial recordings, and where special measures can be taken for the sake of experiment, it is possible to make recordins that are relativ noise free. The comparison of these reorings with most commercial recordings is rather striking: The elimination of the noise has an effect on the listener greater than would be expected considering the small magnitude of noise present in commercial recordings. Progress is being made. The noise level of commercial recordings is being reduced. The last few years have seen a great improvement in magnetic recording tape as well as the development of two effective systems for noise compression during recording. If progress in this direction continues, we may witness the elimination of the problem in this decade.

The problems in recording that are associated with acoustic fields are more formidable and will require significant research. Let's look briefly at one of these problems: the problem of recording a solo violin. As is the case with many musical instruments, the polar pattern (a plot of sound pressure level versus direction of radiation) of the acoustic radiation from the violin becames very complex as the wavelength of the sound becames small compared to the size of the_instrument. Polar patterns for 660 Hz (a violin's open E-string) and 6,000 Hz (the ninth harmonic of open E) are shown in the illustration at the right. Recall that the frequency spectrum of the reverberant field is related to the frequency spectrum of the total power radiated by the acoustical source rather than to the balance of frequencies on any one axis of radiation. In order to determine the total power radiated by the vìolin at 6,000 Hz, it would be neccessary to iniegrate over a series of polar patterns of the type shown in the illustration, each representing the radiation in one plane, until we had spanned a spherical surface with the violin at its center.



The radîation patterns of a violin's open E string (660 Hz; top pattern) and the ninth harmonic of open E (6,000 Hz; bottom pattern). Six dB difference between two measurements corresponds to a doubling of sound pressure. The radiation pattern of the violin becomes complex as the wavelength of the radiation shrinks toward the dimensions of the instrument. (Source: "Effects of Directional Radiation from Violins upon their Recorded Sound," by Victor Nedzelnitsky. Undergraduate thesis, ELectrical Engineering, M.I.T., 1966.)

If we were to place a recording microphone close enough to the violin to be predominantly in the direct field, we see from the polar plot that the sound pressure level picked up by--the microphone could easily vary Iw 20 dB depending upon the position of the microphone. For example, if the microphone were located in a position corresponding to 90° on the 6,000 Hz plot, the sound pressure level would be more than 20 dB greater than the sound pressure level that the microphone would pick up if it were located at an angle of 70°. At other frequencies, the situation is very different. For example, we can see from the polar pattern for 660 Hz that the 70° and 90° angles yield essentially the same sound pressure. It is plain that we cannot expect to find any angle for the microphone placement that will at every frequency yield a signal proportional to the total power radiated at that frequency by the violin. In other words, when the microphone is placed close enough to the violin to be predominantly in the direct field, it does not pick up the spectral balance of the reverberant field that wc experience in the concert hall. Instead, it picks up a balance of frequencies proportional to the radii of the polar patterns at each of the different frequencies for the angle of the microphone position.
The solution to this problem is not so simple as removing the microphone to the reverberant field, because in recording there, we encounter all the normal modes of the recording environment, which, when reproduced through a second set of normal modes-those of the listening room-produce a barrel-like sound that is totally unacceptable. The recording engineer is really caught between two undesirable limits. If he places the microphone close to the performer in the direct field, he picks up a spectral balance that is critically dependent upon microphone position, and that is not the desired balance of the reverberant field. But if, seeking to avoid this problem, he moves the microphone farther away and into the reverberant field, he encounters the normal mode problem.
It is interesting to note that, without realizing the nature of the technology invoived,

most recording engineers usually place their microphones at a distance (when normalized for microphone dirrectionality characteristics) from the performer that is close to the distance at which the levels of the direct and reverberant fields are equal.

In this way, the spectrum of the recording is influenced to some extent by the spectrum of the reverberant field without too heavy an influence of the normal modes.
This represents a good compromise between bad alternatives.

We would like to be able to record in such a way that we capture the spectrum of the total radiated acoustic power from the performer unaltered by the normal modes of the recording environment. We are presently studying the possibilities for designing acoustical environments that lead in this direction. It would not surprise us to find that recordings of the future will be made in unusually shaped environments that act as Il acoustic lenses" to focus the radiated power, and that these recordings will then be fed into computers that will process the signals so that when they are played back in homes they will produce a reasonable sensation of the large environment of a live performance.

In summary of our predictions of the future:

The state of the art in HiFi components is such that, with the exception of dynamic range in recordings, we do not expect major improvements in performance from individual components. The evolution of components will be in reliability, size, weight, and cost.

We expect dramatic improvements in the overall performance of home music systems, but we think they will come through _systems engineering rather than_ through component development. By systems engmeering, we mean engineering that considers the entire system from recording to playback, develops appropriate signal processing techniques, and uses existing components, or modified versions of them, as building blocks to synthesize new and strikingly superior music systems. This is exactly the course the electronics industry has followed. First the individual circuit elements were developed, then circuits were sythesized, and now systems engineering has combined these circuit building blocks to create complete high performance systems in every discipline from industrial process control to guidance systems for space vehicles. The HF industry is now ready for systems engineering.

My profession has habituated me to making homework assignments at the conclusion of presentations. This assignment is lighter than this presentation has been, and it is disgustingly appropriate: Read Hans Christian Anderson's "The Emperor's New Clothes," and substitute "HíFi" for "clothes" throughout. At the conclusion of this assignment you will bave a good picture of much of the "development" and "evaluation" efforts that have characterized the HiFi field. But with a little bit of luck, we may be at a turning point which will see more serious research and consequent progress. At the risk of being proved wrong, let me conclude with a prediction that I must admit is entwined with hope. If research bears the fruits that I believe are possible, the home music system of the future will bring to many people an appreciation of musie unimagined today.