The Reproduction of Sound

In the design of systems to record and replay complex sounds the acoustical engineer seeks not to re-create sounds exactly but to satisfy the peculiar requirements of the ear and brain

by Edward E. David, Jr.

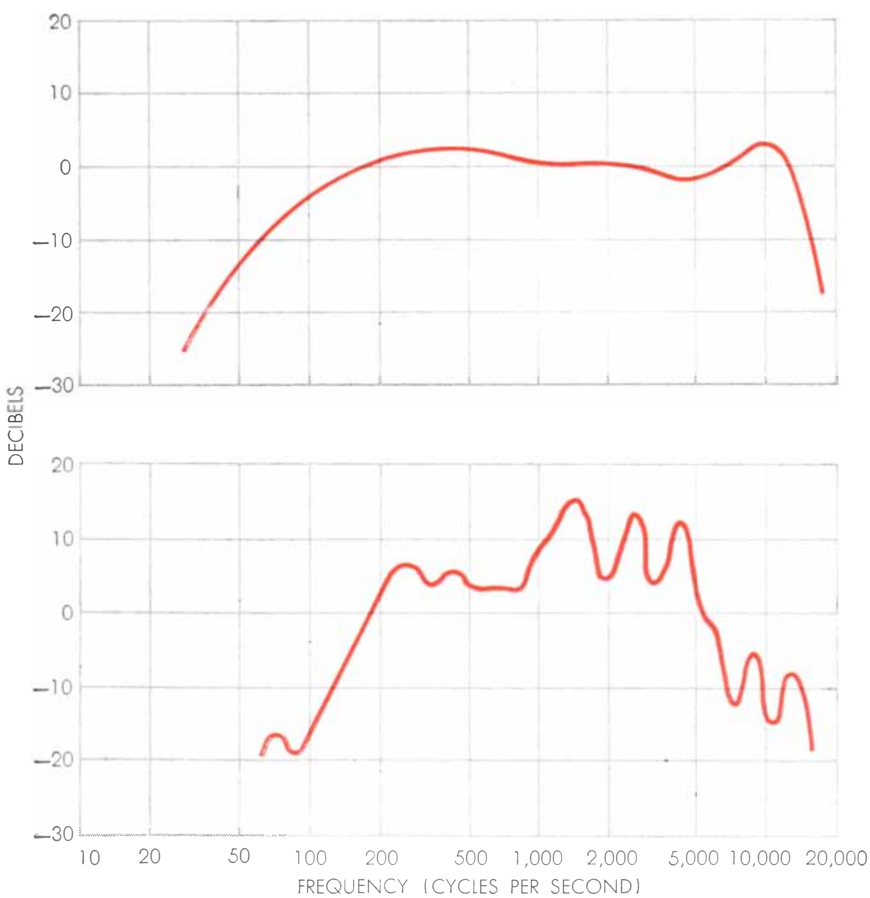
Does a falling tree make a noise when no one is nearby to hear it? This famous question, propounded by the British philosopher Bishop Berkeley, points up a fundamental duality in the concept of sound. A physicist would doubtless answer yes; the crash produces a spreading disturbance in the air—in other words, a wave of sound. A psychologist might say that the disturbance does not become sound until it is perceived.

The acoustical engineer has a foot in each camp. In manipulating sound physically—storing it, transmitting it, projecting it—his eventual aim is to deliver a wave that, in falling on a listener's ears, produces a sensation resembling as closely as possible the sensation that the waves from the original acoustical event would have produced. Fortunately, subjective duplication does not require physical duplication. The human auditory sense is deaf to many gross distortions and omissions. What cannot be perceived need not be preserved. Therefore the engineer does not have to try to duplicate every detail of the original waves when reproducing them. That would be a hopelessly complicated task.

On the other hand, the hearing apparatus is surprisingly sensitive to certain effects. And human preferences often depend as much on experience as on present sensation. The distinction between auditory necessity and irrelevance and the subtle connection between the physics and the aesthetics of sound pervade all of acoustical technology.

Modern sound engineers are expert at turning physical means to psychological ends. Under some conditions even experienced listeners cannot distinguish between the sound produced by the best reproduction systems and that of the original performance. Nevertheless, there is still room for improvement in certain of the standard components. Engineers are only now beginning to learn to use sound systems to tailor the acoustics of rooms, concert halls and theaters. Here too the physical-psychological duality is the key.

Physically sound is created by vibrating bodies that set up traveling condensations and rarefactions in the air or other material around them [*see illustrations on page 74*]. A tuning fork sets up condensations and rarefactions that blend smoothly into each other. Such a sound is a pure, or monochromatic, tone. The spacing between condensations determines the number reaching the ear each second, since they all travel at the same speed in a given medium. When this number, or frequency, lies between



FREQUENCY RESPONSE determines the fidelity of a sound-reproduction system. In a system with good frequency response (*curve at top left*), the piano note middle C (*recorded*

20 and about 16,000 per second, the normal young adult listener can hear the tone. The physical frequency of the tone determines its subjective pitch. The greater the frequency, the higher the pitch. The lowest note on a piano corresponds to the pitch of a 27.5 cycle-per-second tuning fork, whereas the highest corresponds to a frequency of 4,186 cycles per second.

A second subjective characteristic of the tone—its loudness—depends on the physical intensity of the condensations and rarefactions, that is, on the degree of compression and expansion of the air. (The relative intensity of two sound waves is measured in a unit called the decibel. If a particular wave is arbitrarily chosen to represent zero decibels, then a wave at 10 decibels is 10 times as intense, one at 20 decibels 100 times as intense, one at −10 decibels a 10th as intense, and so on.)

Sound Waves

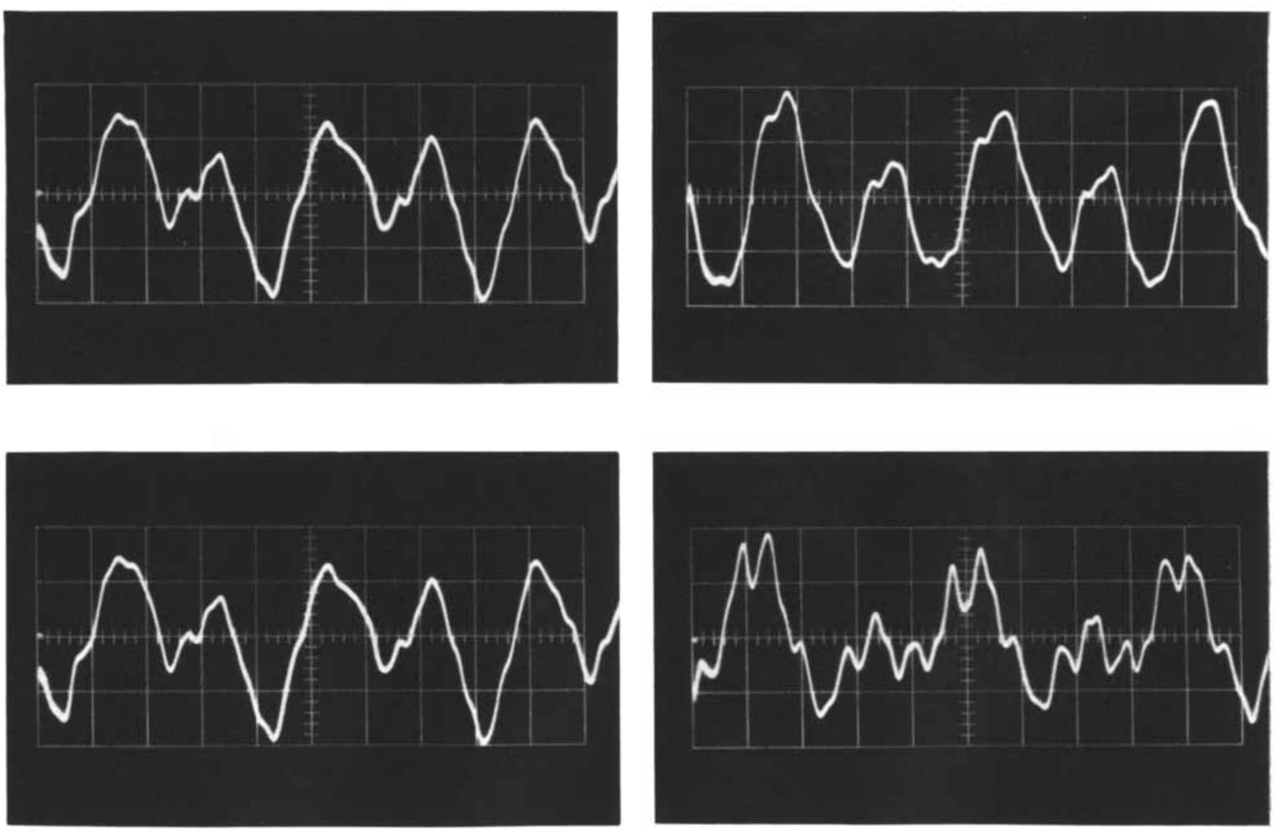
For the purpose of picturing or analyzing sound waves, the direct representation of condensations and rarefactions is not very helpful. It is more convenient to plot a curve showing variations in air pressure, the crests corresponding to points of maximum condensation and the troughs to points of maximum rarefaction. When a pure tone is plotted this way, the curve turns out to be the familiar sine wave.

In practice, of course, practically none of the sounds we hear are pure tones. If the waves were visible, they would not look like the regular processions of compressions shown in the diagrams but would vary from place to place in both spacing and intensity. The pressure curve would no longer be a simple sine wave but would actually have a complex shape.

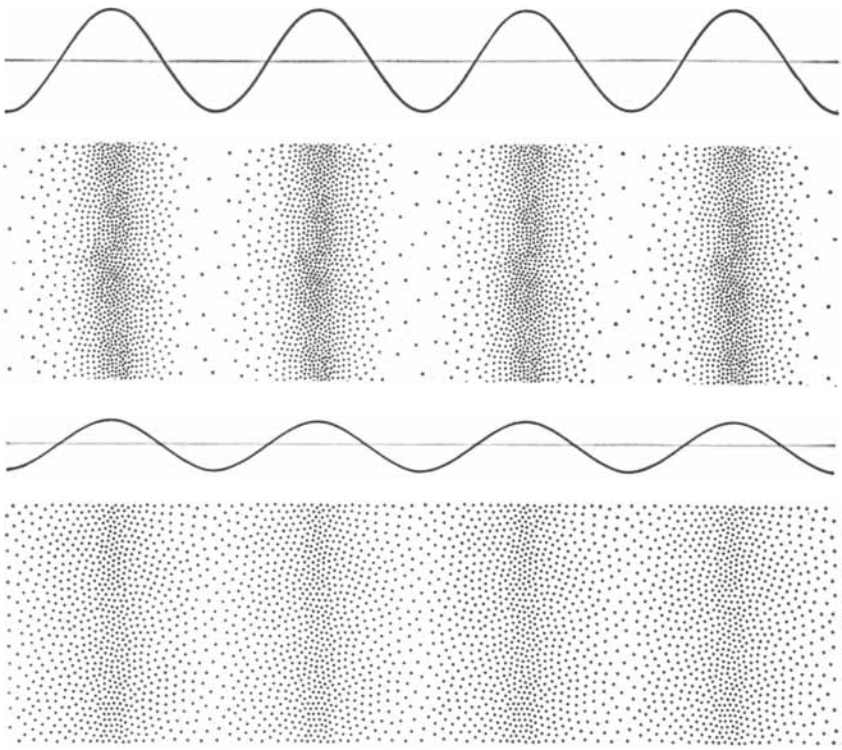
Some 150 years ago the French mathematician Jean Baptiste Joseph Fourier discovered that such shapes, even when extremely complicated, can be formed by adding together various sine waves [*see illustration on page 75*]. This fact implies that every sound, whether a musical note or a discordant noise, is a combination of pure tones. In musical notes the lowest tone establishes the pitch. The remaining sine-wave components, or harmonics, determine the quality, or "tone color." The difference in quality between a French horn and a trumpet, say, when both are sounding the same note, depends on the relative intensities of the harmonics produced by each instrument.

About 50 years later Hermann von Helmholtz found that the subjective quality of a sound depends almost not at all on the relative phases of its harmonics. As the drawings on page 75 show, shifts in phase among sine-wave components can drastically alter the shape of a resultant curve. Yet the ear almost totally disregards the change. It tends to hear each harmonic as a separate tone, combining them without regard to phase. This is a lucky circumstance; it is an exacting and expensive task to build electrical and acoustical circuits that preserve the relative phases of the components in a wave.

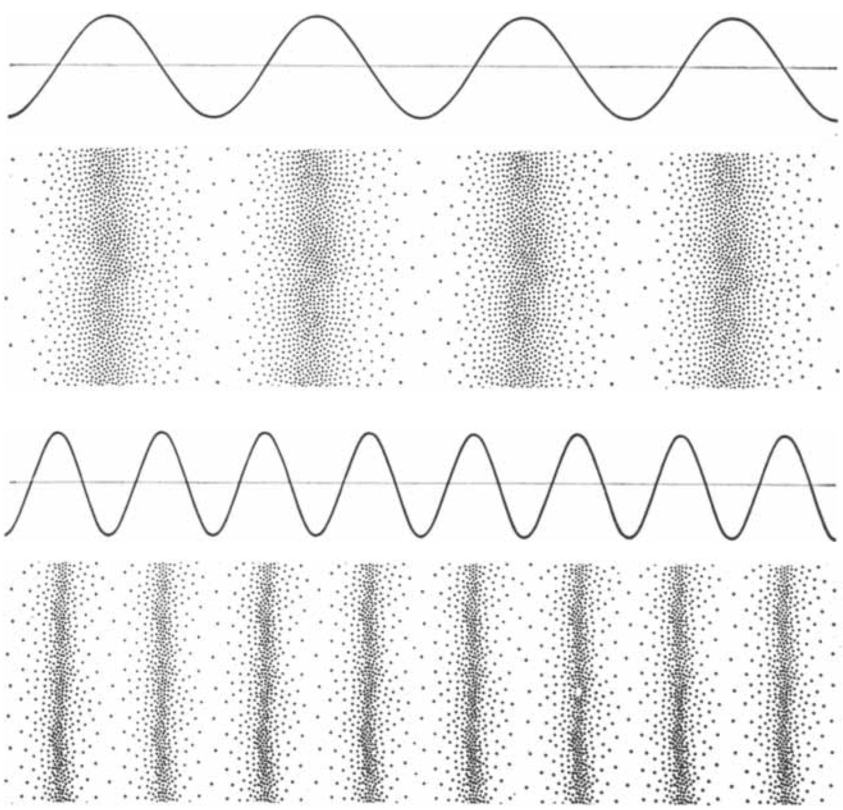
An ideal acoustical system, then, would deliver to the listener a wave containing all the sine waves that were in the original wave, and only those. Their relative intensities would be strictly preserved, but not necessarily their relative



***in oscillograph at top center*) is reproduced with high fidelity (*oscillograph at top right*). In a system with poor frequency response (*curve at bottom left*), the same note (*bottom center*) is reproduced with low fidelity, *i.e.*, considerable distortion (*bottom right*).**



INTENSITY of sound waves (*sine curves*) in air (*dots*) depends on the degree of condensation (*closely spaced dots*) and rarefaction (*widely spaced dots*). Degree of condensation and rarefaction in wave at top is approximately twice that in the wave at bottom.



FREQUENCY of sound waves (*sine curves*), measured in cycles per second, is the number of condensations or rarefactions that pass a given point in one second. The frequency of the wave at bottom is twice that of the wave at top. Both waves are equal in amplitude.

phases. Needless to say, there are no ideal systems.

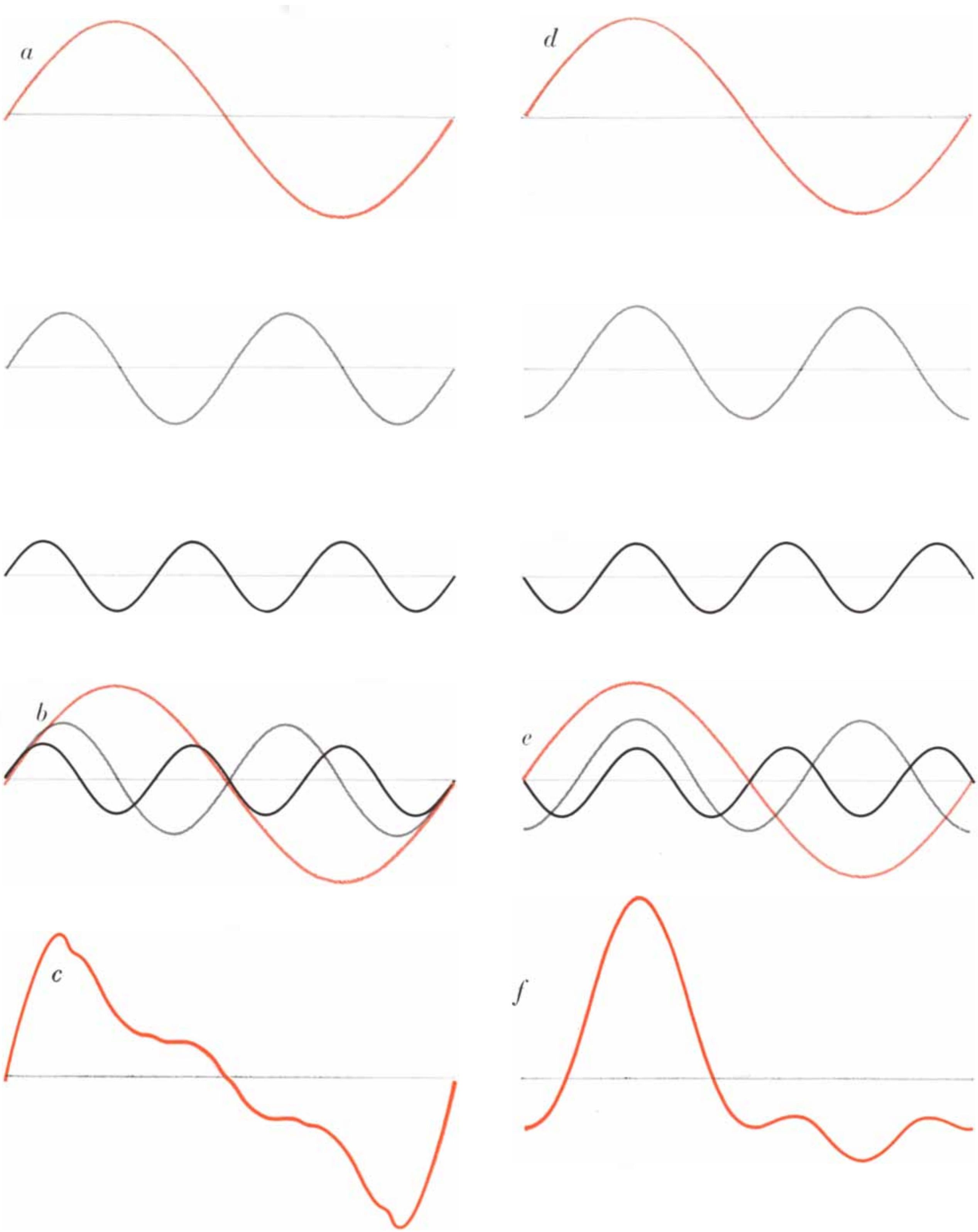
How do they fall short? They may degrade sounds in two general ways. First, an imperfect frequency response may overemphasize the intensities of some sine waves in the input while depressing or eliminating others. Second, new sine waves, not contained originally in the input, may appear at the output if a system changes the input wave shape. This effect is known as nonlinear distortion.

A system with poor frequency response imposes its own characteristic tone color on any sound passing through it. Cutting off low frequencies gives a thin, shrill sound; high frequencies, a muffled sound. Complex, uneven responses produce distortions that are harder to describe but easy to recognize. The added tone color remains constant, whereas the tone color of the desired sound—for example, orchestral music—is continually changing. Listening to such a system is like looking at a movie through an imperfect pane of glass that distorts all images in the same way. To reproduce tone color with high fidelity a sound system must have a "flat" response, varying no more than three or four decibels, for frequencies from about 30 to 15,000 cycles per second.

Although people are growing accustomed to high-fidelity reproduction and are coming more and more to demand it, particularly in phonographs, they can be, and often are, satisfied with much less. It is not hard to make an unconscious correction for a constant distortion. A listener willingly accepts a tone quality from a car radio or pocket-size portable radio that he would not tolerate from a phonograph. The frequency response of the telephone hardly extends below 200 or above 3,600 cycles per second. To be sure, no one would choose to listen to music over such a system, yet in spite of the lack of low and very high frequencies it serves well for personal voice communication.

Similarly, added components of sound will be discounted if they are not too loud and if they are steady. A little hum from an amplifier or hiss from a record surface can be ignored. Even a mixture of two entirely dissimilar types of sound may be admissible, as in the case of speech and background music. Each has its own tempo, pitch, frequency spectrum and intensity, and the listener has no trouble keeping the two apart.

The most objectionable change that a reproduction system can impose on a sound is to introduce extraneous com-

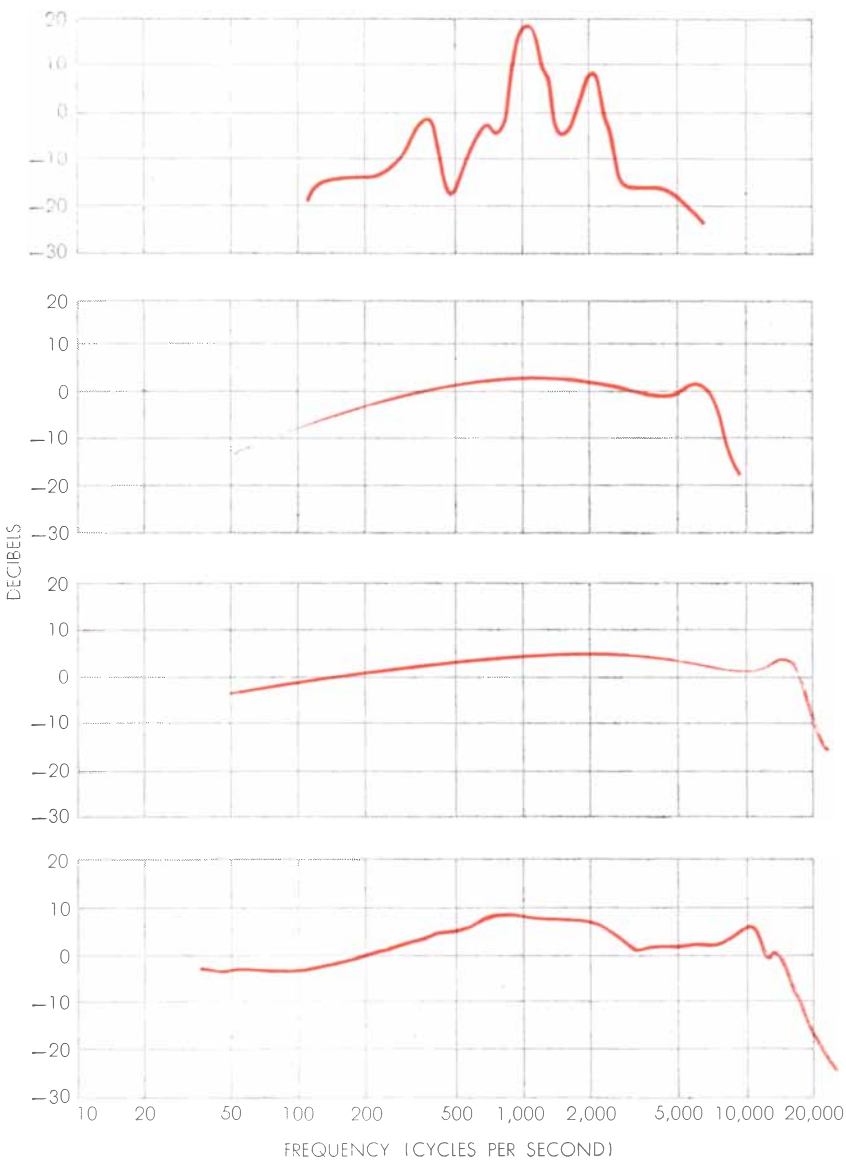


MUSICAL NOTE can be represented by a complex curve (*c*) that is a combination of pure tones: a fundamental (*sine curve at top in* "*a*") and one or more overtones (*two lower curves in* "*a*"), which are heard simultaneously (*b*). When the phases of these tones are shifted with respect to one another ("*d*" *and* "*e*"), the complex curve representing the note is greatly changed (*f*). The ear largely ignores this change, however, and the note produced sounds almost exactly like the previous one (*c*), when there was no phase shift.

ponents with a rhythm of variation similar to that of the original sound. This nonlinear distortion occurs most commonly when the original sound becomes too intense for the capabilities of the system. In physical terms, the input waves are larger than can be duplicated by some component of the reproduction device. As they pass through, their tops are clipped off and they come out flattened. Such flat-topped wave shapes are, as Fourier showed, composed of sine waves, but many components that were not in the original sound are required to produce the shape. In other words, when the reproduction equipment is driven too hard, it adds new components to the sound wave.

The effect is not constant. Once the intensity limits have been reached, the greater the intensity of a wave, the more it is flattened. Therefore nonlinear distortion varies with the intensity of the input sound. In an orchestral piece, for instance, the intensity changes from note to note, and the fluctuating distortion becomes entwined with the music.

In the late 1940's Harry F. Olson of the RCA Laboratories tested the effects of nonlinear distortion on over-all subjective reaction to reproduced sound. He played music to a group of listeners through a sound system with a variable frequency response and variable nonlinear distortion. The wider the range of frequencies reproduced, the less nonlinear distortion the subjects would tolerate. For instance, with the system adjusted to pass only the frequencies between 50 and 5,000 cycles per second, they put up with three times the distortion they would accept when the response was extended to 15,000 cycles per second. This subjective interaction between frequency range and nonlinear distortion constantly plagues the sound engineer. He may take great pains to extend the frequency range of his equipment only to find that he has revealed previously unnoticed faults in the apparatus or recordings.



PERFORMANCE of modern sound-reproduction systems, as measured by frequency response (*colored curves*), has been greatly improved over that of earlier systems. The early mechanical recorders (*top*) responded very unevenly to a narrow range of frequencies. The Maxfield-Harrison recorder of 1925 (*second from top*) showed a smoother response to a still narrow frequency range. Except for the loudspeaker (*bottom*), components in modern systems (*third from top*) show an almost "flat" response up to at least 15,000 cycles per second.

The Origins of Sound Reproduction

The modern technology of sound reproduction is rooted in the work of three American inventors: Alexander Graham Bell, Thomas Edison and the late Lee De Forest. In 1876 Bell's telephone showed that the vibrations of a sound wave could be picked up by a diaphragm and then reconverted to sound by making it set up vibrations in a second diaphragm in an earphone. A decade later Edison's gramophone demonstrated that the vibrations could be permanently stored in the form of a "hill and dale" track cut into a smooth surface. In 1906 De Forest built the audion, a three-element vacuum tube that can amplify electrical signals.

It was almost 20 years more before the three innovations were brought together. The first sound-storage systems were entirely mechanical. The tiny forces in the condensations and rarefactions of sound waves were transmitted by means of a diaphragm and a series of levers to a needle-like stylus, which cut a track in a wax cylinder or disk. The energy came entirely from the sound wave. To obtain enough power for cutting the wax, sound had to be funneled into the diaphragm with a large horn. In reproduction all the energy for the loudspeaker diaphragm was supplied by the needle that retraced the up-and-down track of the record.

The greatest problem that the early acoustical engineer faced was simply to make the sound loud enough. Part of the solution was to design a mechanical system that resonated strongly to a narrow band of frequencies. The result was a narrow and spectacularly uneven response that had, in addition, a generous dose of nonlinear distortion [*see top graph at left*]. Little wonder that early gramophone music sounded more like a product of the device than of the performing instruments.

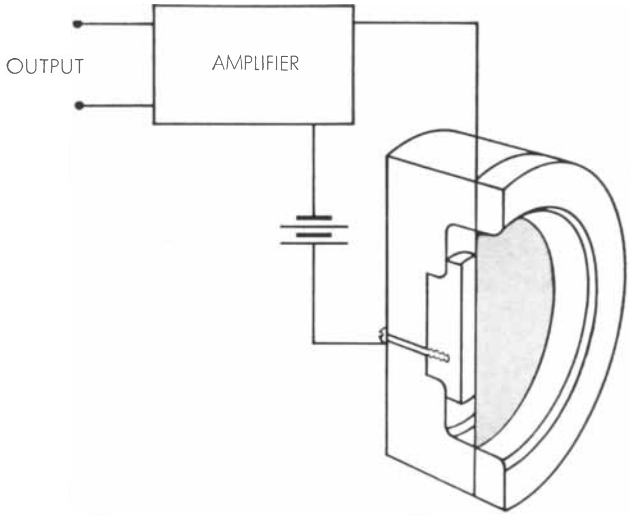
In 1925 Joseph P. Maxfield and Henry C. Harrison of the Western Electric Company took advantage of what was already at hand to create a new technology. They hooked up a microphone and an amplifier to drive an electrically actuated recording stylus and were at once able to improve dramatically on the frequency response of the mechanical recorder. As soon as their "electrical transcriptions" were available it became worthwhile to build an electronic reproducer, with electromechanical pickup, amplifier and loudspeaker.

Maxfield and Harrison's electromechanical recorder is based on a magnetic principle. A tiny slug of iron is suspended between the poles of a permanent magnet with a wire coil wound around it [*see illustration at bottom left on next page*]. The slug is attached to one end of a shaft and the stylus to the other. Fluctuating current supplied to the coil varies the magnetism acting on the slug, making the slug rock back and forth against the force of a retaining spring. The stylus moves from side to side in response to the current in the coil. Thus the sound track is cut laterally rather than up and down. The same device can recover an electrical signal from the sound track. Motion of the stylus tip causes a varying magnetic field, inducing a corresponding voltage in the coil.

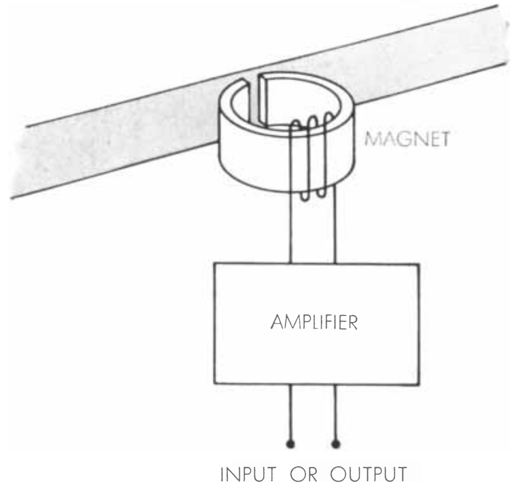
With power no longer a concern, engineers could now turn to the problem of fidelity. The easiest part of the job turned out to be the electronics. In a few years they learned to make amplifiers that far outperformed the other parts of the system. Today it is no trick to build an amplifier that is, within the capabilities of the human ear, nearly perfect. Modern electromechanical recorders and pickups have also been brought to a high level of performance. Present limitations are set by the storage medium itself and the "transducers," which convert between electricity and sound.

Disk, Wire and Tape

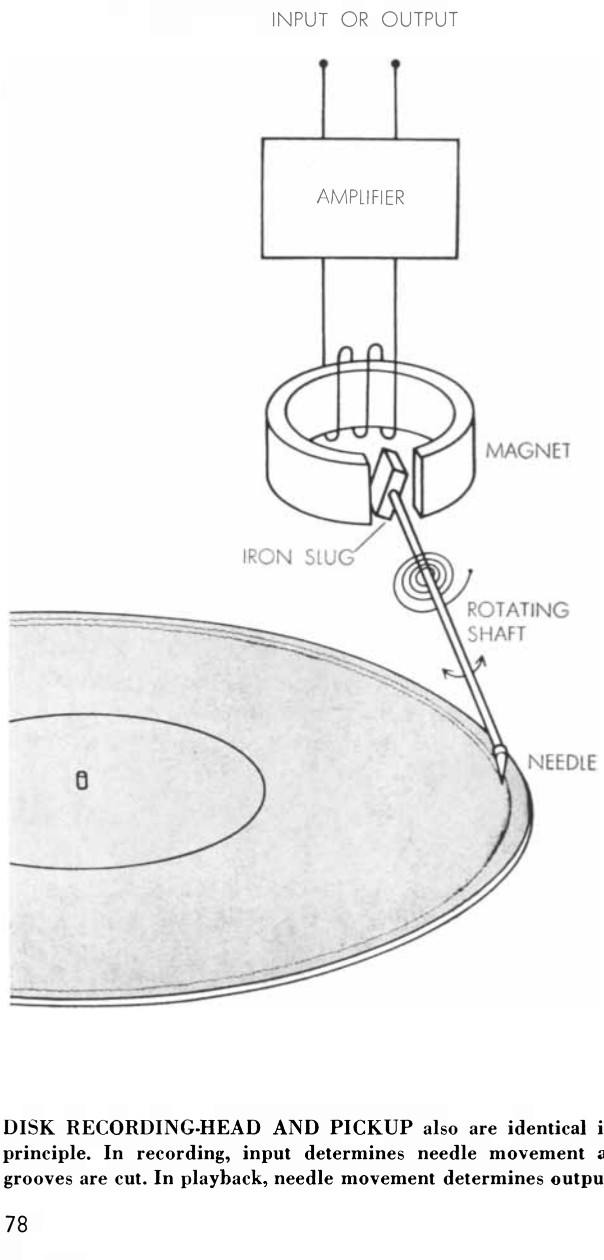
Everyone is aware of the enormous superiority of present disk recordings over those of even 10 years ago. Today's best records are fine indeed but they are still noticeably far from ideal. Their chief drawback is nonlinear distortion from several causes. Some of these are economic. For instance, frugal duplicating procedures may set limits on the fidelity of the reproduced grooves. Other limits are inherent, for instance distortions introduced by the curvature of the



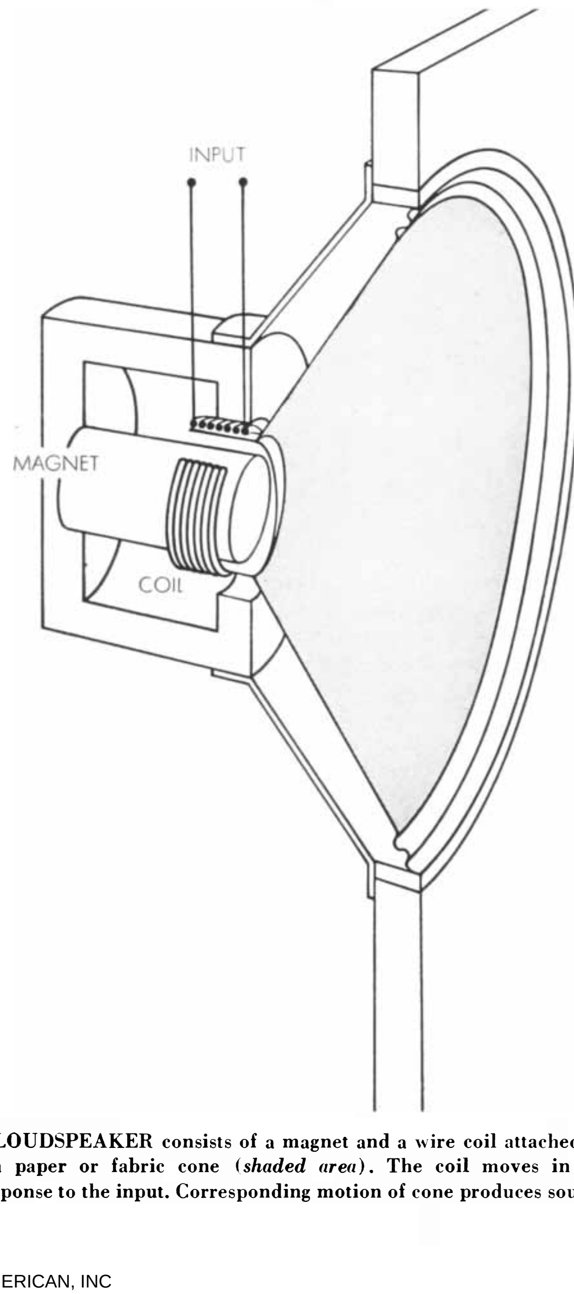
CONDENSER MICROPHONE converts sound into an electrical output. Diaphragm (*shaded area*) in front of condenser plate moves in response to sound waves, varying the gap and thus the output.



TAPE RECORDING-HEAD AND PICKUP are identical in principle. In recording, magnetization of tape (*shaded area*) varies with electrical input; in playback, output varies with magnetization.



DISK RECORDING-HEAD AND PICKUP also are identical in principle. In recording, input determines needle movement as grooves are cut. In playback, needle movement determines output.



LOUDSPEAKER consists of a magnet and a wire coil attached to a paper or fabric cone (*shaded area*). The coil moves in response to the input. Corresponding motion of cone produces sound.

sound track. Then there is the question of durability in use. Great care is necessary if the track is not to be worn out of shape, adding new, unwanted sine waves.

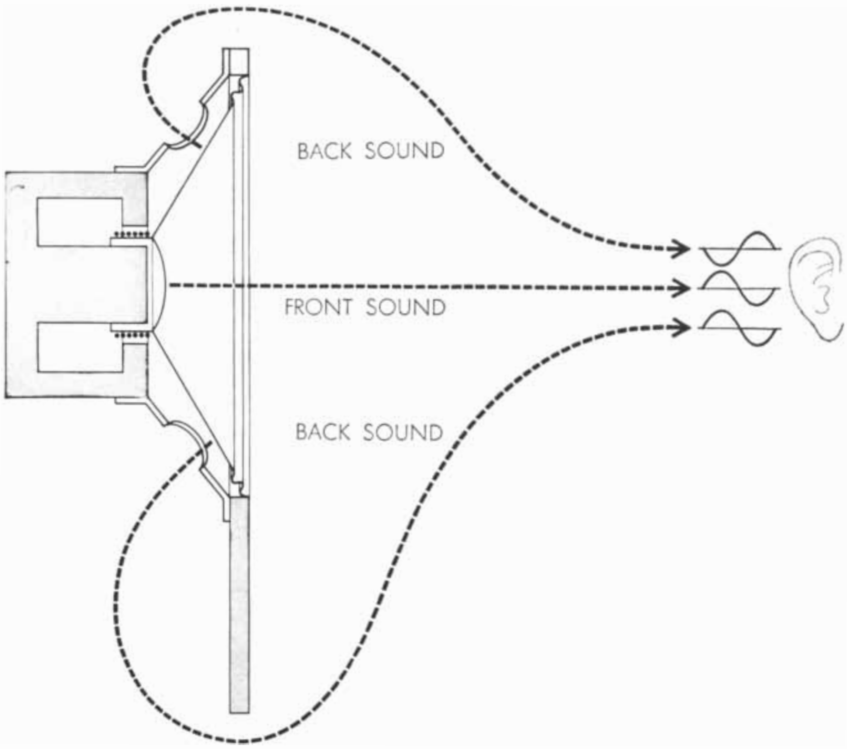
Another shortcoming of the disk is its inability to accommodate anywhere near the full range of intensity in orchestral music. The limits are set at the upper end by the overload point, at which the recorder and pickup begin to respond in a nonlinear manner; at the lower end, by the surface noise inherent in the disk material. The best records have a range of about 40 decibels from the noise level to the overload level (an intensity ratio of 10,000 to 1). At a live performance of a symphony the sound level may extend over 80 decibels. To reduce nonlinear distortion the sound engineer usually cuts the volume of the louder passages before their vibrations reach the stylus. If all the other parts of a system are as good as they can be, the disk record will set the limit on the fidelity.

For the storage of sound, magnetic tape recording was an advance nearly as revolutionary as the coming of electrical recording. As long ago as 1898 Valdemar Poulsen—the "Danish Edison"—discovered how to record sound magnetically. He sent the varying current from a microphone through the coils of an electromagnet and passed a steel wire across the poles. Successive portions of the wire were magnetized with a strength corresponding to the changing strength of the current. When the wire was moved past the poles of a second electromagnet, the changing field induced a varying current in the coil. Sending the current through a headphone reproduced the sound.

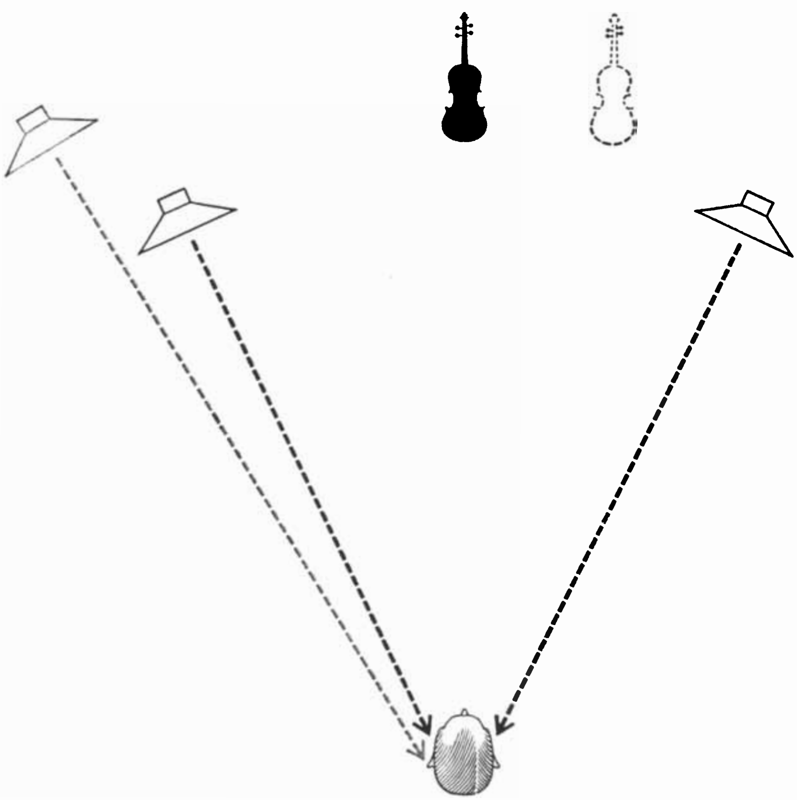
Poulsen actually built a wire recorder called the telegraphone. In the 1920's, when amplifiers had come into use, the device was revived in Germany. Wire was expensive and difficult to handle, however, and magnetic recording did not come into its own until F. Pfleumer, a German engineer, developed plastic tape coated with powdered magnetic material. Before World War II a tape recorder much like those of today had been built in Germany.

Magnetic tape is now the best medium for storing sound. Its intensity range reaches as high as 65 decibels. This is still short of the full symphonic demand but it may be extended with further research. The limits are set by the range of maximum possible variation in magnetic strength of the tape—from the noise level to the overload point. Improvements in the magnetic materials and techniques of fabrication should make it possible to operate over a wider range.

Although wear is less of a problem



CONDENSATIONS of low-frequency sound waves from rear of a loudspeaker ("*back sound*" *at top*) tend to cancel rarefactions simultaneously produced at the front ("*front sound*"). A baffle (*shaded area at bottom*) delays back sound, thereby reducing interference.



SOUND OF VIOLIN from two symmetrically placed loudspeakers is perceived by a listener as coming from a violin (*black*) midway between them. If one loudspeaker (*gray*) is moved away, the listener locates the source (*dashed violin*) nearer the other loudspeaker.

than it is with disks, tape recordings are not permanent. They may deteriorate in storage when temperature and humidity are not carefully controlled. Moreover, each layer on the reel tends to "print through" to the next, resulting in an annoying "pre-echo." No solution for the latter problem is in sight.

The Transducers

Other critical points of the sound-reproduction chain lie at the junctions between sound and electricity. At the microphone end, high-quality instruments have been available for many years. For example, the principle of the condenser microphone dates from before 1880. The microphone is an electrical condenser consisting of a thin diaphragm mounted in front of a rigid plate [*see illustration at top left on page 78*]. A constant electric charge is maintained on the plate. As the diaphragm vibrates under the influence of impinging sound waves, the spacing between it and the plate changes, changing the capacitance of the instrument and therefore the voltage between the plates. The small voltage variation is fed into an amplifier. In the early days the condenser microphone had to be abandoned because, without amplification, it produced too little power. Now it is one of the strongest links in the chain of sound reproduction.

At the other end of the chain is a weak link—the loudspeaker. The principle of operation of the modern device was discovered in 1874 by Werner von Siemens, founder of the German electrical equipment firm Siemens & Halske. Siemens placed a coil of wire in a magnetic field directed radially outward from the coil axis and passed an alternating current through the wire. As the field of the coil interacted with the permanent field, the coil moved back and forth along its axis.

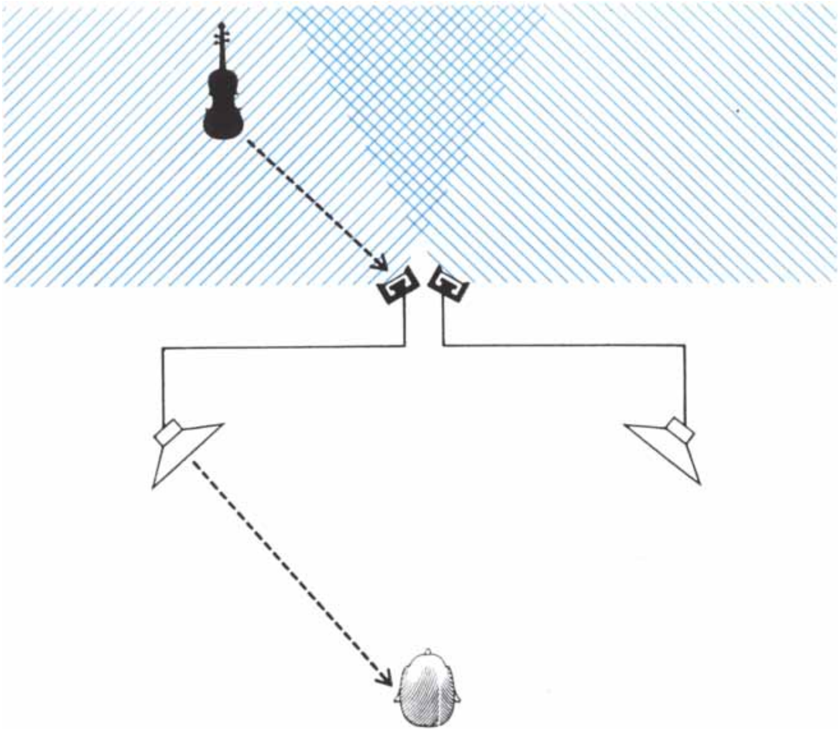
Modern electrodynamic speakers embody Siemens' principle in the "voice coil," which receives the fluctuating electrical output of an audio amplifier. As the coil oscillates it communicates the motion to a flaring cone of paper or fabric and thence to the air. In the early days great difficulties were encountered in achieving a uniform frequency response over a wide range. In the 1920's C. W. Rice and E. W. Kellogg of the General Electric Company analyzed the mechanism and found that the coil and magnet respond less efficiently to the electrical signal as its frequency increases. By proper design, however, this effect can be offset by an increase in the efficiency with which the cone sets up air vibrations. This is still the basic design principle of cone loudspeakers.

In analyzing the action of the speaker cone, Rice and Kellogg found that the surface should be large for good low-frequency reproduction and small for high frequencies, a conflict that still plagues designers. One compromise makes use of a cone that is stiffer near the center than it is near the outer edge. At low frequencies the cone vibrates as a unit; at high frequencies the stiff inner portion moves almost independently of the flabby rim. A second way out is separate speakers—small ones for the highs and large ones for the lows. Although multiple speakers are now the preferred solution, they still present difficulties in achieving a smooth response at the frequencies of crossover from one speaker to another.

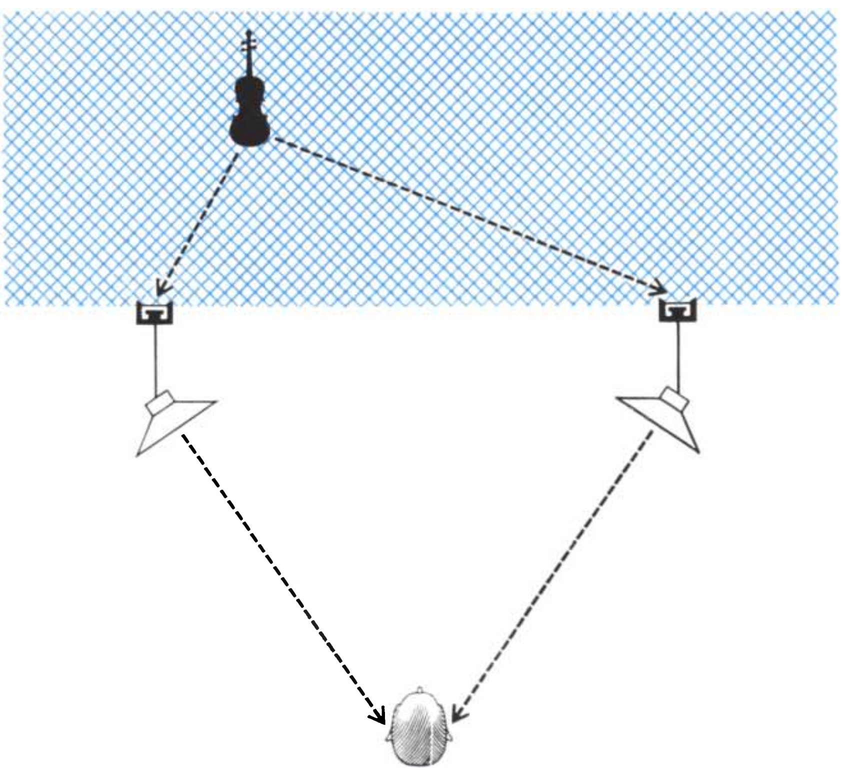
Designers of loudspeakers face other troubles. One arises from "back sound": each time the front surface of the cone creates a compression, the back surface creates a rarefaction and vice versa. Some of the energy from the back wave reaches the front of the speaker, after traveling a longer distance than the direct wave [*see top illustration on opposite page*]. At low frequencies, where the wavelength is measured in feet, the extra distance is not enough to shift the relative phase of the two waves substantially, and the back sound tends to cancel the front sound. Below a certain frequency, therefore, the speaker muzzles itself.

One answer to the problem is to surround the speaker with a baffle, which increases the length of the path of the back sound and reduces the frequency at which it begins to interfere. This notion was carried to its logical conclusion in the "infinite baffle," which took the form of a large box lined with absorbent material that completely soaked up the back radiation. This radiation can, however, be turned to useful account in the "bass reflex" enclosure. Here the speaker is mounted in a box that has an open port in the front. A box of proper dimensions resonates at low frequencies and emits radiation from the opening that reinforces the direct sound. Still other schemes lead the back sound through a labyrinth so that it reaches the front in an advantageous phase.

The horn loudspeaker provides an even better solution. Here the cone or a small piston is placed in the throat of a long horn. This allows the moving surface to drive the air efficiently and uniformly over a wide range of frequencies and reduces the back radiation. Horns are large and unwieldy, however, even



STEREOPHONIC SYSTEM of A. D. Blumlein is based on directional microphones and separated loudspeakers, which reproduce sound from source at center (*crosshatched area*) with equal intensity. Loudspeaker at left reproduces violin sound with greater intensity.



SEPARATED MICROPHONES connected to separated loudspeakers is basis of stereophonic system developed by Harvey Fletcher. Sound coming from source nearer one microphone is reproduced earlier and sometimes louder in the corresponding loudspeaker.

when folded up and placed in a corner of a room.

These shortcomings imply that loudspeakers offer a great opportunity for further improvement. By the same token, the purchaser of a sound system will do well not to stint on his speaker, since it is likely to set the limit on the quality of the set.

Perspective and Ambience

Even an ideal loudspeaker at the end of a chain of ideal components could not reproduce for a listener the sensation he experiences in a concert hall. Sound has spatial as well as tonal quality. At the concert the listener has auditory "perspective"; he can hear that the violins are on the left, the cellos on the right and the soloist in the middle. He also hears sound from many directions other than from the stage—diffuse reverberations from walls and ceiling, curtains and carpets. The impression produced by this diffuse sound is called ambience. Both perspective and ambience disappear almost entirely when sound is projected from a single speaker. It is like listening to a concert from the lobby of a concert hall through a chink in the door to the auditorium.

The ability to locate a source of sound, which is responsible for the sense of perspective, depends on hearing with two ears. Sound from any direction other than straight ahead or behind arrives at one ear a little earlier than at the other and, because of the shadowing effect of the head, a little louder. Somewhere in the brain the signals from each ear fuse into a single, composite image. The minute time and intensity differences between them serve to fix the location of the source.

Ambience arises when many successive echoes of a sound arrive at the listener's ears from different directions. Again the brain combines the complex train of events into a unified impression of the size, shape and texture of a room or hall.

Any attempt to reproduce perspective and ambience must provide something different for each of the listener's ears. As long ago as 1881 a Frenchman, Clement Ader, demonstrated a system in which sound from two spaced microphones was piped separately, via telephone lines, to a corresponding pair of earphones. In the 1920's it was found that the perspective effects from such a binaural arrangement were enhanced by putting the microphones on each side of a dummy head. Binaural systems provided a startlingly realistic impression, except in one respect: when the listener turns his head, the sound rotates with him—he is always facing the orchestra. Besides, the headphones are a nuisance. For both reasons the binaural scheme has now given way to stereophonic systems, in which the two earphones are replaced by loudspeakers fed by independent microphones.

Each speaker sends a sound wave to both ears of a listener, but he perceives a single image. If the two sources are equidistant from the listener and are equally loud, he hears the sound as coming from halfway between them. If one speaker is moved a little closer, so that its sound arrives a little earlier, the image shifts toward it. This phenomenon is known as the precedence effect: the first sound to arrive from an acoustical event pre-empts the location mechanism. An interval of about five-thousandths of a second between the two sounds is enough to shift the apparent source all the way to the earlier speaker. The same result is obtained by making the sound from one speaker louder than that from the other. A ratio of 10 decibels moves the image to the louder speaker. For larger differences in delay or intensity the image stays at the earlier or louder speaker.

Although these effects are the source of the stereophonic illusion, such systems had been developed long before the principles were understood. In the 1920's an ingenious British inventor, A. D. Blumlein, Harvey Fletcher of the Bell Telephone Laboratories and K. de Boer and A. van Urk of the Dutch firm N. V. Philips' Gloeilampenfabrieken laid the groundwork for stereophonic reproduction. They hit on the same basic plan: two separate sound channels from microphone to speaker. The schemes differed in the way in which they picked up the sound.

Blumlein used the equivalent of a pair of directional microphones, placed close together with their axes of maximum sensitivity at right angles to each other [*see top illustration on opposite page*]. In this arrangement a sound originating off the center line gives a louder response in one microphone and in the corresponding speaker. The system operates on the intensity effect described above. Fletcher's setup, on the other hand, consisted of a pair of nondirectional microphones spaced several feet apart [*bottom illustration on opposite page*]. Here the precedence effect is applied: sound from an off-center source reaches one microphone sooner than it reaches the other and therefore emerges first from the associated speaker.

Both approaches are in current use; Europeans generally prefer the Blumlein system and U.S. engineers the Fletcher system. While they rely on different physical principles, their psychological effects are very similar. The closely spaced microphones do give more consistent reverberant patterns, however. When the microphones are widely separated, the echoes arriving at each one may be quite dissimilar. Moreover, the echoes can vary drastically with the progress of the music since echoes from various surfaces may reinforce each other at some wavelengths and interfere at others. In extreme cases an instrument such as a piano will seem to "ping-pong" from side to side as different notes are sounded. Such undesirable effects can be largely controlled by proper studio design and careful placement of the microphones.

In both systems the auditory perspective changes if the listener moves off the center line between the two speakers. Most people do not mind the distortion; the exact location of each instrument is not important. As a matter of fact, many listeners feel that the principal advantage of stereo lies not in the localization of particular sources of sound but in the ambience that the system affords. The somewhat different echo patterns picked up by the two microphones and projected by the speakers combine at the listener's ears to call up an impression of a diffuse field of sound coming from the general region of the speakers.

Recently some investigators have used twin speakers to produce ambience from a single microphone or recording track without regard to the spatial separation of sound sources. M. R. Schroeder of the Bell Telephone Laboratories has devised a system that generates multiple echoes, simulating reverberation. The sound is continually recorded on a rotating magnetic drum, picked up at a later point in the rotation and fed back into the system. Two delay units supply different echo patterns to two speakers, which give the impression of a diffuse source of sound. Many listeners find this "quasi stereo" system hardly distinguishable from the real thing and some even prefer it. The system is no simpler or cheaper than stereo, but it does project single-channel recordings much more satisfactorily than one speaker can. It is particularly helpful in improving the sound of old recordings with limited frequency range and considerable distortion.

A Dutch engineer, Roelof Vermeulen, has carried quasi stereo a step further in a technique he calls ambiophony. He places speakers around the listener (or listeners): at the sides and in the rear of the audience as well as in front. The speakers at each location are fed through delays that allow the direct sound to reach that point before the corresponding sound emerges from the speaker. The result is an all-around system of artificial echoes that resemble those in a concert hall. Increasing the delays to the speakers increases the apparent size of the enclosure. Ambiophony can turn a living room into the acoustical equivalent of an auditorium. Artificial reverberation is the only way to augment the direct sound in outdoor concerts, and it can enhance the acoustical properties of real concert halls and auditoriums.

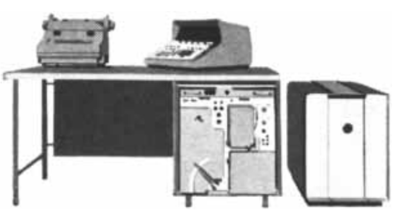
A similar scheme can also remedy the effects of too much reverberation. Prolonged reverberation severely reduces the intelligibility of speech because each sound "rings" for several seconds, masking what comes after. Railroad stations and churches are traditional offenders. Halls designed primarily for music, therefore, are often not suitable for speech. Excessive reverberation can be offset by increasing the direct sound through directional loudspeakers, which beam their output to the audience. The additional direct sound adds little reverberation of its own since most of it is absorbed, principally by the clothing of the audience. If the reinforcing sound arrives somewhat later than that from the person speaking, the precedence effect hides the electroacoustic system.

Through these techniques the acoustics of an auditorium can be tailored to each occasion. Both St. Paul's Cathedral in London and the Palais de Chaillot in Paris employ directional speakers to increase the intelligibility of sermons and plays. The musical acoustics in the famous Teatro della Scala in Milan have been markedly improved by an ambiophonic arrangement. The Grand Auditorium at the 1958 Brussels Exhibition contained a system permitting flexible control of both direct and reverberant sound. Plays, speeches, orchestral and choral concerts, chamber music, solo recitals and film presentations were each accommodated with appropriate acoustical support. In this country there is still prejudice against electroacoustic sound reinforcement but it is dying out. In the future electroacoustic systems will doubtless become an integral part of the design of theaters and auditoriums instead of an afterthought. Indeed, the fusion of electroacoustics with architectural acoustics is the new frontier in sound engineering.

ANOTHER IN THE CAUTIONARY SERIES ABOUT

RECOMP II [in which we reveal nearly all the computer's subtle enticements]

Here, for the very first time, is an unexpurgated listing of the allurements of Recomp II. They are potent. They are persuasive. They are enticing. Indeed, in reading them it would be wise to exercise a decent restraint... for you may find yourself falling in love with a computer:



1] The *first* solid-state digital computer on the market was Recomp II. The *finest* computer on the market is still Recomp II. Recomp II's dedicated engineers are determined that the situation shall remain this way; they have an obsessive regard for this precise machine they have so carefully developed.

2] Recomp II is the *only* compact computer with built-in floating point arithmetic. It defies being hemmed in on a problem. With its large capacity it obviates computer-claustrophobia.

3] Always the darling of the medium-scale computer user, Recomp II has been so well accepted that it can now be offered at a *significantly lower price.* It still provides the identical quality, solid-state performance, and features that can't be found on computers costing three times what Recomp II *used* to cost.

4] Recomp's memory, employing the new optional Recomp MagneticTapeTransport units, would stagger an elephant. Each unit has a memory of over 600,000 words. Up to eight of these transport units can be connected to Recomp II, giving you a computer with a total memory capacity of over 5,000,000 words. Remarkable!

5] The speed of the new magnetic tape control described above is quite remarkable, too. Read and write speed is 1850 characters a second; bidirectional search speed is 55 inches per second.

6] Always the darling of the medium-scale computer user, Recomp II has been so well accepted that it can now be offered at a *significantly lower price.* It still provides the identical quality, solid state performance, etc.

7] Recomp II seems to have more built-in features than a dream-home kitchen. It has built-in square root command. Built-in automatic conversion from decimal to binary.

8] Would you say that a RUG is important to a computer? Well, it is. The RUG we mean is RecompUsers Group. This is a highly active group, sharing up-to-date information, and keeping in close contact by monthly bulletins. A consideration not to be overlooked when buying a computer.

9] Always the darling of the medium-scale computer user, Recomp II has been so well accepted that it can now be offered at a *significantly lower price.* It still provides, etc., etc.

10] Recomp's keyboard, which you can see above, looks easy to operate. It is. And because it requires no specialized talents, anyone with computer problems can be taught to use Recomp II. Quite a handy machine to have around.

11] Recomp II can easily be installed anywhere, requiring no more electricity than an ordinary electric toaster.

12] Another new optional feature for your Recomp II is the Facitape tape punch and reader console, shown below. It punches 150 characters a second, reads 600 characters a second, and stops on a character. It adjusts to read and punch from 5 through 8 channels. Quite uncanny.

13] A visual readout on the control panel allows you to check any information about to be entered into Recomp II before you press the "enter" button. The information can be corrected easily if necessary. This is further evidence of Recomp II's staunch adherence to efficiency.

14] Recomp II's programming is highly efficient. It has 49 basic instructions, expandable to 72. Word length is 40 binary bits; very large, indeed.

15] Always the darling of the medium-scale computer user, etc., etc., etc.

16] Recomp II has a large sub-routine and program library, and a large program exchange.

17] Each word programmed into Recomp II contains two instructions.

Recomp II has many other features, but as you can see, space is running short. We would have liked to have lingered upon the details of Recomp II's own full scale compiler called SALT, and even maybe discuss the high-speed loops a little . . . but. Perhaps, if you are beginning to feel the stirrings of your acquisitive instinct toward Recomp II, you should see it in action. We can arrange a demonstration for you through our local offices in New York, Chicago, Boston, San Francisco, and Long Beach. Or, at the very least write for more information. We have some nice brochures you will enjoy reading.

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