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STEREOPHONIC SOUND

by Norman H. Crowhurst

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STEREOPHONIC SOUND

NORMAN H. CROWHURST

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To my daughter: *DEIRDRE MICHELE*

PREFACE

Everything these days is a pursuit. Automobiles go after more horsepower. High fidelity amplifiers were going after more watts, until they ran out of bigger tubes. Movies went after bigger screens. And when they rolled back the curtains on these super-wide screens, they needed sound that "rolled back" with them. Thus stereophonic sound came to the movies.

But it did not begin with the motion picture industry. Believe it or not, the first experiments in this direction were demonstrated at the Paris Exposition of 1881! But naturally it had to wait for the advent of high fidelity before stereophonic sound could really come into its own, which it is only recently beginning to do.

This subject seems to have grown up in a sort of no man's land. Ask an engineer, "What is stereophonic sound — how is it different from high fidelity?" and he will probably launch into higher mathematics to explain it to you. Ask a musician the same question, and he will weave a lot of artistry around his explanation, that may mean something to him . . . So who *is* to tell us about it?

Maybe I am a citizen of this no man's land. While various loudspeaker enthusiasts have, in almost reverent tones, chanted praises to such mystifying terms as "back-loaded horn", "bass-reflex", "infinite baffle", "acoustic suspension", I have tried to keep myself informed on the relative merits of these wonderful developments, without joining any individual camp of absolute worshippers. The same goes for people with such very definite ideas on other subjects — the "right kind" of phono pickup, for example. And, although I was trained as an engineer, I have endeavored to understand the musician's viewpoint — the artistic aspects, and at least I have, to show for this effort, a passing acquaintance with the piano keyboard!

Do I know the answers, and have I told all there is to know? Everyone "in the business" will tell you there is still a lot to learn, and you ask me if I know it all! But let me say this: the information I have presented here is based on progressive experience in the subject's development. I have listened, investigated, compared notes with the experiences of many others, technical and otherwise, and they all seem to add up, consistently, to the answers I have given in this book.

Don't let anyone tell you that only an expert can understand the intricacies of stereo. Sometimes the inhibitions of "experts" makes *them* slower to perceive the simpler things than ordinary folks, and nowhere is this more true than in stereo. The endeavor here has been to convey to you, the reader, what I have learned, in such a way that you will be able to confirm it with your own experience, and thus grasp the subject more readily. Further, by giving you the pros and cons along the line – where there are such – you will be set to go on learning from your experience in this intriguing and highly controversial subject, long after your first reading of this book.

A good stereophonic system, with well-recorded program material for it, can give the most satisfying experience. So here's hoping this book will lead you on, without unnecessarily depleting your budget, to some solid listening satisfaction.

NORMAN H. CROWHURST, M.I.R.E., A.M.I.E.E.

New York, N. Y.
September 1957

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Chapter 1

BINAURAL LISTENING

No doubt about it, big words like binaural and stereophonic can be impressive, particularly when used by the man who *sells* stereophonic sound. Then you get the impression that if you only have ordinary, un-stereophonic sound, you haven't lived yet. So, "All right", you say, "I'll have stereophonic sound", but then you discover that the cost of both equipment and recorded material for stereophonic sound is about twice that of ordinary, un-stereophonic sound.

This makes you think again; "Just what do I get for all this extra cost? In what way does stereophonic sound differ from un-stereophonic sound?" The salesman has an answer ready. Most probably he will draw your attention to the difference between ordinary photographs, either black and white or in color, and 3-D or stereoscopic pictures. He will explain to you that ordinary photographs are two-dimensional, or flat, while stereoscopic pictures provide you with the third dimension, making the pictures appear in "depth", which is what we mean by calling them 3-D.

This is a wonderful made-to-order illustration for the stereophonic salesman, because the difference between two-dimensional and three-dimensional pictures is certainly dramatic. He will tell you that the difference between two-dimensional and three-dimensional sound is equally dramatic. He may then play you a carefully selected and prepared demonstration that he hopes will persuade you into parting with just twice as many dollars as you would for your ordinary un-stereophonic sound system.

In all fairness however, we must admit that while there are plenty of salesmen of this type around, there are also many who will give you a fair demonstration and explanation. The situation remains somewhat unsatisfactory though, because a great many people are out of reach of qualified advisers who could give a lucid explanation of just what stereophonic sound is. And that is precisely why this book has been written.

That rather obvious illustration of two-dimensional and three-dimensional pictures, while dramatic, is not quite a true parallel for the difference between ordinary and stereophonic sound. In the realm of sound there is no such thing as a direct counterpart for the two-dimensional or flat picture. It is quite impossible to have a two-dimensional sound. I am not saying there is no difference between stereophonic and un-stereophonic sound. There is. But the difference is not to be so easily explained as by drawing an illustration with 2-D and 3-D pictures.

The qualities about sound and listening that give us an impression of depth are quite different from the qualities about light and vision that give us a similar impression of the things we see. That is why this book begins with a chapter on binaural listening: with the object of establishing a clear picture of just how we hear things. Let's just go around in our everyday experience and notice some of the things that our hearing can do for us.

Most people who read this will have two good ears and the remarks that follow are primarily for their benefit. If you happen to be partially deaf in one ear you will probably find that most of the remarks mean something to you, while some of them are a little difficult to verify in your case.

If you are stone deaf in one ear you may still have some sense of direction, but it will be quite different from that described in the rest of this chapter, and what you read will therefore just be theory – it will have little or no meaning in your experience.

Learning to Hear

Just how good is our sense of direction in hearing? Have you ever tried calling a child's name, or for that matter an animal, and seeing how accurately he turns his head to look directly where you are? At first, when very young, the child or the animal may need to do a little searching, after looking in the general direction, before he finally sees you. But as his hearing becomes better trained his head will turn in the precise direction in which you happen to be.

This lesson is learned of course, with the aid of vision, but when hearing reaches this degree of precision the fact that it is not dependent upon vision any more can be verified by hiding behind suitable obstacles such as a convenient tree or other hiding place. Although he can no longer see you when he turns his head in that direction, you will find that his hearing direction perception is so acute that he still looks in the correct direction. He can probably identify immediately just which obstacle of many you happen to be hiding behind.

This sense of direction is not confined to sounds coming from a horizontal plane or on a level with you. This you will be able to observe by listening to aircraft going overhead, if you are in the middle of an open space, such as a field, or beach, or a large parking lot. You will find that the first notification you usually get of the approach of an aircraft is its sound. If you are interested to see the aircraft, your head then turns to look in the direction from which the sound is coming and you quickly find it with your eyes.

Sound Takes Time to Travel

If the airplane is one of those slow-flying kites from a near-by airstrip, the position your ears find will be so close to that where you see the airplane that the sound seems to come right from the aircraft itself. If it is a commercial airliner with a cruising speed of 250 to 300 miles per hour, then the general sense of direction will be similar, though you may notice that the sound does not seem to be so closely identified with the aircraft as it did with the slow-flying job.

But if a modern jet comes overhead at a speed much nearer the speed of sound, say 500 or 600 miles per hour, your sense of hearing alone will not help you much in finding the aircraft visually, until you realize that you have to look some distance ahead of where the aircraft sounds to be. The sound of the aircraft will seem to be traveling in a certain direction, and by looking considerably further ahead in that direction you will find the aircraft itself. This is because between the time the sound left the aircraft and reached your ears, the aircraft itself had traveled a considerable distance (Fig. 1).

From this kind of listening you will realize that trained hearing can identify directions of sound to within a few degrees. If you happen to be listening in city streets, however, you may find some difficulty. The aircraft may not be in the direction it seems to be at all, because the sound is reflected from the walls of the buildings that line the street. This is because sound waves can be reflected and a false sense of direction consequently given. If a number of

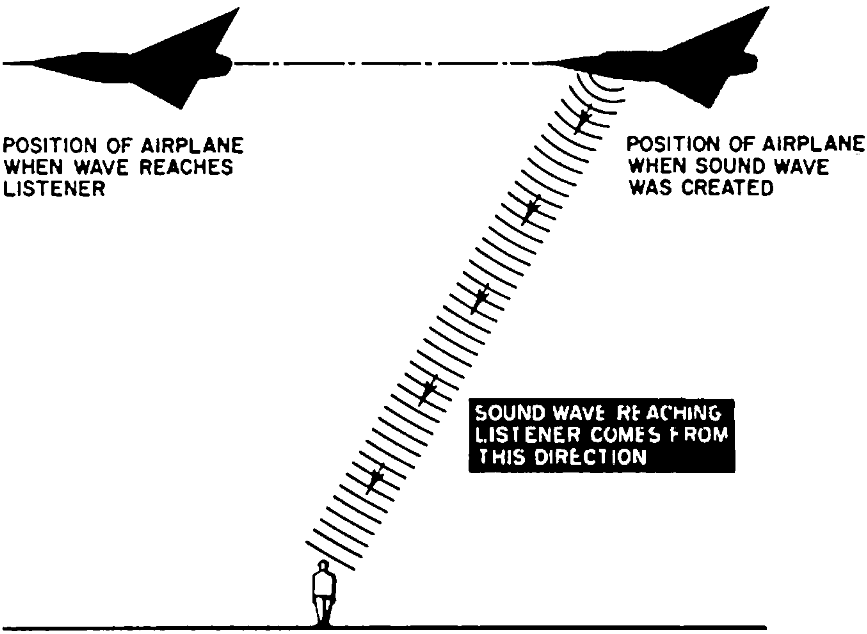


Fig. 1. Because sound takes time to travel, the audible position of a fast moving aircraft is "behind" its actual position.

reflections occur at once all sense of direction is lost and the sound becomes confused.

However, sounds do not have to be outdoors for our sense of direction to operate. If someone speaks to us indoors we can just as readily turn to look in the direction their voice comes from, as if they speak to us outdoors. Notice too that you can very easily locate a clock by the sound of its ticking.

But not all sounds can be so readily located. For example, have you ever been in a gas lit auditorium where one of the gas jets whistled, and tried to locate which jet the whistle came from? You can walk around the building and get no positive identification of direction at all. Eventually the simplest way seems to be to turn off the gas jets one by one until the whistle disappears.

A more modern illustration of this point might be a building heated by steam radiators, where one of the radiators emits a whistle. From the sound you might think that all the radiators whistled, but by turning the radiators off one at a time you will probably find that the whistle emanates from just one of them. Your sense of hearing however, gives you little help in finding which one.

How is it that the source of direction of some sounds can be so readily identified, while others seem to be so confusing? The reason for this sheds considerable light on what we need to get effective stereophonic sound, so we shall discuss it at greater length later on in the book. Meanwhile let's pursue a little further the things our binaural, or two-eared, listening enables us to do.

Selective Hearing

So far we have concentrated on obtaining a sense of direction from the sounds we hear. But our hearing faculties can differentiate on other bases than just direction. Have you ever talked to someone in a crowded room, where many other conversations were taking place at the same time? Next time you're in this position, try relaxing your attention for a moment.

You will realize that the background of conversation going on is such as to be completely confusing, unless you concentrate on listening to the person with whom you are talking. Your hearing faculty enables you to single out your friend's voice from all the other voices in the room. If you listen carefully you will realize that this is achieved by three separate factors acting together:

(a) Your sense of direction in hearing is concentrated toward your friend's voice;

(b) Your eyes may be helping you to some extent by a degree of lip reading; and finally –

(c) Your hearing faculty provides a kind of selective quality whereby your friend's voice is separated out, by his own personal individuality, from all the other voices.

For another example of the power of the hearing faculty to discriminate between sounds, consider listening to an orchestral concert, either live, or reproduced over a really high quality reproducer. You can quite readily identify sound from different instruments in the orchestra. They stand out clearly, each with its own individual characteristic sound.

Even at a live concert, from where you are sitting in an auditorium, you may have difficulty in locating the position of the violins by the sense of sound alone because of reverberation in the building. But you will have no difficulty in distinguishing the sound produced by the violins from sounds produced by other instruments, such as wood-wind or brass.

Although it utilizes the same range of component vibration frequencies that other instruments use, each kind of instrument has its own characteristic sound. An analysis of the frequencies used

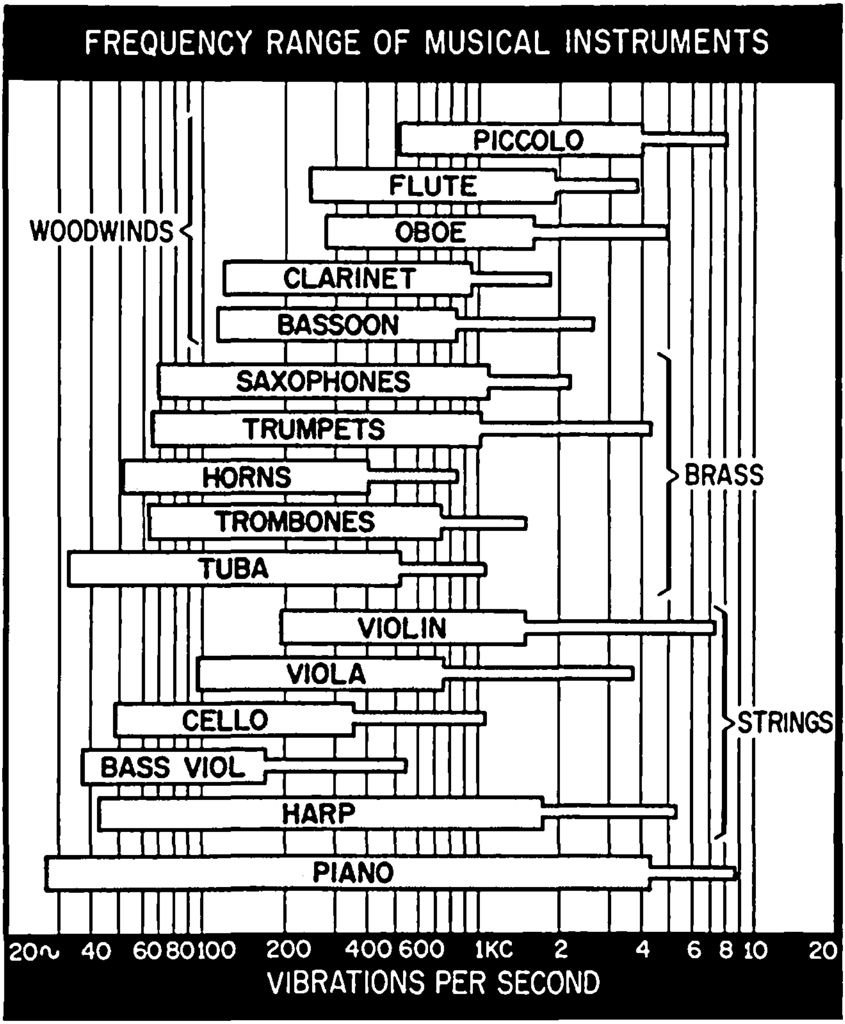


Fig. 2. The frequency range used by different musical instruments overlaps almost completely, so we cannot separate the sound from individual instruments by distinguishing their frequencies.

by different instruments (Fig. 2) shows that it would be impossible to isolate sounds made by one instrument, or kind of instrument, from all other sounds, merely by selecting certain frequencies. Somehow the hearing faculty must identify the *composite* sound from whichever instrument attention is given to.

Another example of extreme selective listening is the auto mechanic who can listen to the composite of weird mechanical sounds that come from an automobile engine, isolate any particular group coming from the valve mechanism, the crankshaft, or whatever part he is giving his attention to, and locate the fault quite readily. To the untrained ear the sound may be almost identical with that from any other motor of the same type; the little sound the mechanic hears may seem inaudible.

Reverberation

These are all observations from everyday experience that any reader can verify for himself. But there is another kind of separation that takes place. We have all subconsciously experienced it, but it is a little difficult to demonstrate directly. This is the separation of original sound from reverberation, which is another name for sound produced by reflection or echo effects.

In a large, reverberant auditorium, the echo is quite noticeable, separate from the sound that causes it. Outdoors, where there may be an echo rock effect, this is also true. But in the average livingroom, or in smaller auditoriums where the acoustics are reasonably good, the reverberation is not noticeable consciously as a separate entity. Our hearing faculty does however, subconsciously separate the original sound from the reverberation.

This can be indirectly but effectively demonstrated with the aid of a good quality tape-recorder, a microphone with good frequency characteristics and some musical friends. Get the friends to perform on one side of your livingroom (Fig. 3), place a microphone approximately in the middle and listen on the opposite side while they do so. You will have an impression of the musical program without any noticeable reverberation. The room is much too small for that.

But now make a recording and play it back. The amount of reverberation you will find on the recording has to be heard to be believed. It sounds as if someone had not only removed all the furniture, but also made the room considerably larger, so as to get definitely more echo.

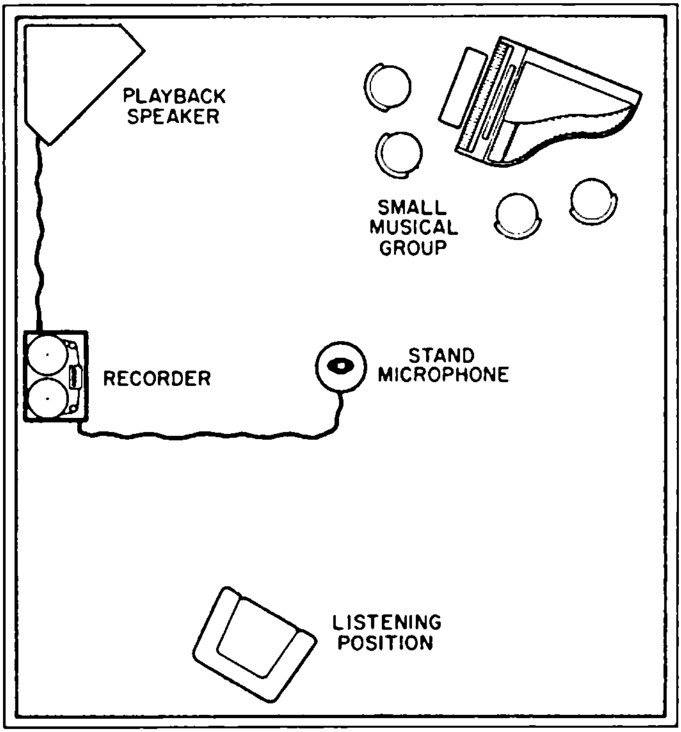
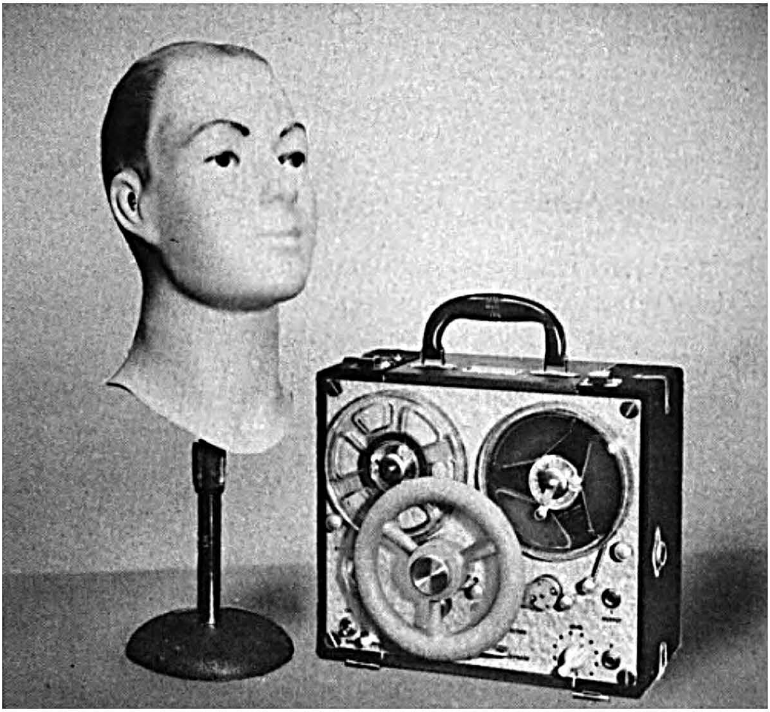


Fig. 3. This experiment (see text) demonstrates that our binaural hearing faculty subconsciously "ignores" reverberation in normal listening conditions, something no reproducing system, even stereophonic, can do.

The reason for this difference is that the one microphone has to receive all the sounds, original and reverberant, and it is reproduced over one loudspeaker. The original sound you heard by means of *two* ears, which enable you to hear it subconsciously distinct from its reverberation. The hearing faculty then, because this is a characteristic of the room that is always with you, subconsciously reduces the noticeability of the reverberation.

The microphone has no means of doing this, and once the program has been recorded and reproduced your hearing no longer has the means to do it either, because the sound, which was previously interpreted separately by each ear, has now become all mixed up.

Much of the skill with which our hearing faculty can discriminate between different sounds, arises from the use of two ears to pick up separate samples of the complete sound pattern and pass them



Courtesy Amplifier Corp. of America

Fig. 4. A portable recorder with dummy head to hold two microphones, for making home binaural recordings.

on to the brain for a sort of computer analysis, which we subconsciously perform all the time.

Referring back, if instead of just one tape-recorder you get a twin-track recorder, and two microphones spaced apart about the distance between an average pair of ears – with some kind of obstacle between the microphones to represent an average head – and then make a recording, you will be able to reproduce this so it sounds like the original program. One way of getting the right pickup is to use a dummy head, with microphones mounted inside (Fig. 4).

Reproduction is achieved by playing back each recording into one of a pair of headphones so that each ear of the listener hears what was received by the microphone corresponding to that ear in the original performance. Although each channel in the tape-recording has just as much reverberation recorded along with it as did the original single-channel recording, we have now provided the hearing faculty with the means for achieving satisfactory separation. Listening to both at once, the reverberation no longer sounds excessive but seems about natural, as it did when listening to the original sound.

Binaural Reproduction

Here, then, we have the original basis for the whole idea of binaural listening and stereophonic sound. Binaural reproduction, as it is called, uses a pair of headphones and a dummy head, as just described. But the method has two principal shortcomings. Although the program reproduced over a good pair of headphones, gives a much better sense of realism in this particular respect, it lacks realism in two others.

Firstly, when the sound originates from a pair of headphones the ear does not behave in the same way, or give the same impression to the brain; one rather has the impression of the sound being "piped" to each ear instead of the ears being free to pick up sounds out of space. In addition to this effect on perception, there is the discomfort of wearing headphones throughout an entire performance.

The second major shortcoming arises from the fact that the recording is made with the head of the dummy in a fixed position — the microphones and the dummy head to which they are attached do not move the way the listener is free to move his head — whereas if you move your head *at all* while listening to the program the effect you get is that of the entire auditorium with its program rotating along with your head (Fig. 5). This is quite unnatural, and the only way to avoid it seems to be to wear a kind of head clamp or have the headphones clamped to the back of the seat in which the listener sits.

Because of these drawbacks of binaural reproduction using headphones, it seems necessary, to get a practical sense of realism, to use different loudspeakers and generate a complete wave pattern in the room. If the loudspeakers could be so placed that each ear of the listener heard only the sound coming from the loudspeaker intended for that ear, the problem would be solved. Unfortunately there is no such simple solution.

Once the sound is released into the room by the loudspeaker, the sound from that loudspeaker travels all around the room. So the sound intended for each ear travels all around the room too, and both ears hear something of the sound intended for the other

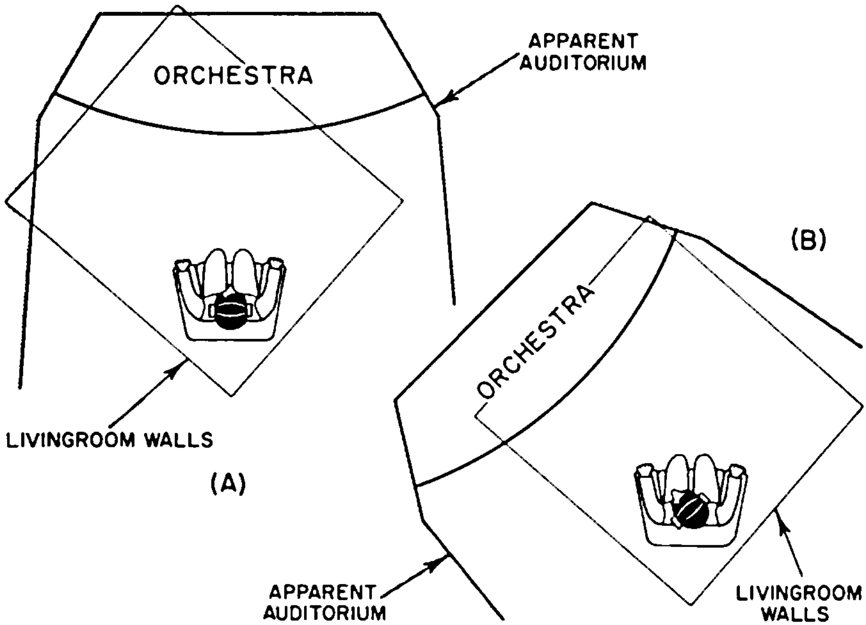


Fig. 5. One disadvantage of binaural reproduction with headphones is that if you move your head the whole apparent program source, realistic as it may be, appears to turn with your head.

ear, as well that recorded specifically for itself. This problem and how it is overcome however, we will discuss later in the book.

For the time being, before we leave the subject of binaural listening, it is important to understand a little better how the hearing mechanism operates to give us the impressions we have noted so far in this chapter.

How Human Hearing Works

Audible vibrations, which we call sound, are fluctuations in air pressure at a rate or frequency which determines the pitch or tone of the sound heard. The lowest audible frequency is in the region of 20 cycles or vibrations per second, while the highest is somewhere around 16,000 cycles. These figures are average for a number of people; yours may vary from these figures. The lower limit may be between 18 and 22 cycles, while the upper limit varies, with individuals, from 8000 to 20,000 cycles.

The fluctuation in air pressure for sound of average intensity is not more than about 1 millionth of the actual pressure of the air. This means the ear has to be extremely sensitive to minute pressure variations within this frequency range.

First of all, the sound vibrations, all of them, pass along the outer tube from the ear to the ear drum (Fig. 6). This is an extremely thin, stretched membrane which vibrates along with the air in contact with it. On the inside of the ear drum is a small air cavity, the pressure of which is equalized with the outside air by

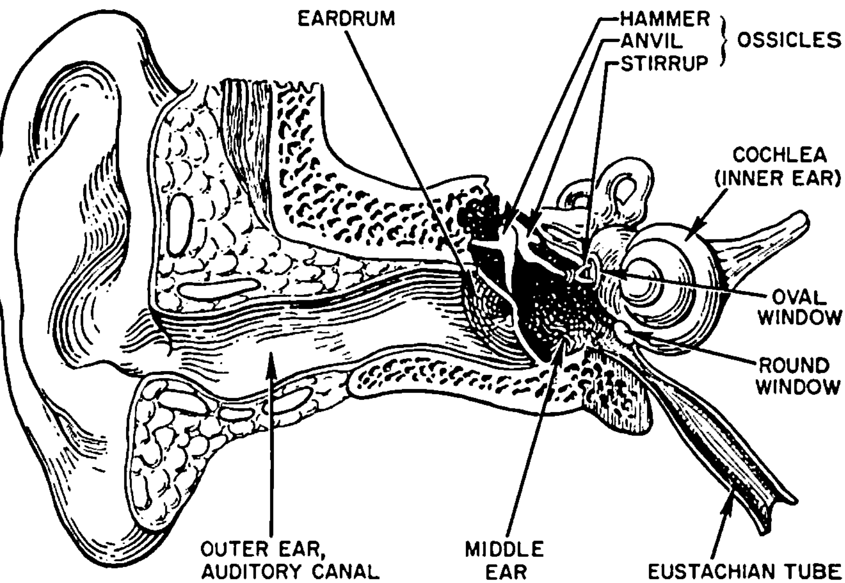


Fig. 6. Construction of the outer and middle ear.

a very small tube called the Eustachian tube, that connects to the outer air again through the mouth. The narrowness of the tube and the fact that it communicates with the mouth, prevents any of the sound energy from bypassing the ear drum, and ensures that the air vibrations we call sound waves cause the ear drum to vibrate in accordance.

This vibration is next communicated to the liquid inside the cochlea (inner ear) by a sequence of three very tiny bones, or ossicles, called the hammer, anvil and stirrup, inside the cavity called the middle ear. The purpose of these bones is to convey all the vibrations of the ear drum, which come from air waves, to vibrations of the oval window, which sets up waves in the liquid contained in the cochlea.

This liquid is contained in two long channels "wound" into the form of a spiral, and separated by a thin membrane known as the basilar membrane (Fig. 7). The whole make-up of the cochlea is contained in a hard, bony, snail-like structure, so that bodily movement of liquid at the oval window, caused by vibrations transmitted from the stirrup bone, must eventually produce corresponding movements at the round window, in the opposite direction.

The liquid may move all the way up the basilar membrane and back down the other side, or it may move only a short way up one side of the membrane before pushing the membrane aside, thus passing the motion to the liquid on the other side.

At just what point it causes the basilar membrane to vibrate depends on the frequency of vibration: the higher the frequency, the shorter distance the liquid moves before it causes the membrane to vibrate; the lower the frequency, the further up the membrane will be the point of vibration.

The place where the membrane vibrates causes stimulation of the nerve cells associated with it, and this in turn is carried to the brain by one of the nerve fibers in the auditory nerve. The auditory nerve has between 20,000 and 29,000 ganglion nerve fibers, each of which carries "information" about one particular frequency of sound being heard.

Careful research into the mode of transmission along the nerves between the ear and the brain shows that this information is passed along in exactly the same way as the nerves in the rest of the human body work, by a system of pulses.

Human nerves do not transmit continuous electric currents, but a succession of pulses, each of uniform size. The brain interprets all information received along the sensory nerves by analyzing how many pulses are received and noting along what fibers they arrive

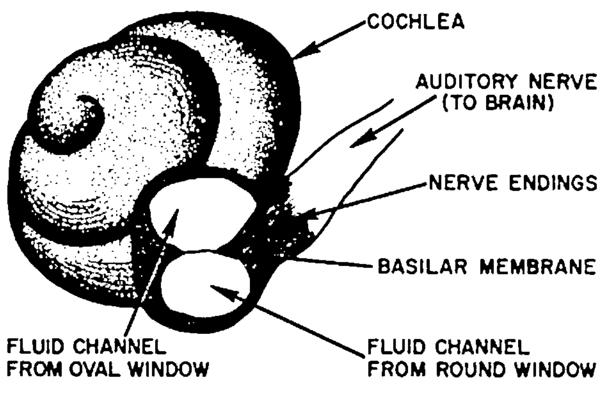


Fig. 7
Cross-section through part of the cochlea, or inner ear.

— a combination pattern, or immeasurably superior morse code or teletype system.

A loud sound does not produce a *stronger* impulse along the appropriate nerves than a soft sound; it produces a greater number of pulses. If the vibration is too feeble to stimulate even one nerve pulse, then it is not audible. This determines what is called the "threshold of audibility". When a sound has become audible it is able to produce one pulse along the nerve about every fortieth of a second — to give the impression of the quietest possible continuous sound.

Further increase in its intensity or loudness will eventually cause a second pulse to be transmitted in an interval less than one fortieth of a second. The rapidity with which the pulses are transmitted along the nerve fiber representing a particular frequency, indicates to the brain how loud that frequency is (Fig. 8).

The auditory nerve terminates in a section of the brain similar to that where all other sensory nerves terminate. The nerves tell what is going on around us. In the same way that the pattern of nerve impulses received from our fingers can tell us the shape and texture of an object we may be feeling, the pattern of pulses received along the auditory nerve can tell us all we want to know about sound we may be hearing.

The grouping of individual nerve fibers along which impulses arrive at the same instant, together with the way in which the impulses speed up or slow down to indicate the change in intensity of different components of the tone, tell us what kind of instrument or sound we may be listening to. One grouping indicates a vibrating string; another indicates a wind instrument, where the tone comes from air particles vibrating in the mouth of a horn or an organ pipe.

The way the individual component tones vary, causing difference in the pulse rate, tells us whether the string has been plucked, bowed or struck. We could go on describing the different qualities about sound that can be identified by listening, but the possible variety in the way impulses can arrive over some 20,000 to 29,000 nerve fibers, is virtually infinite.

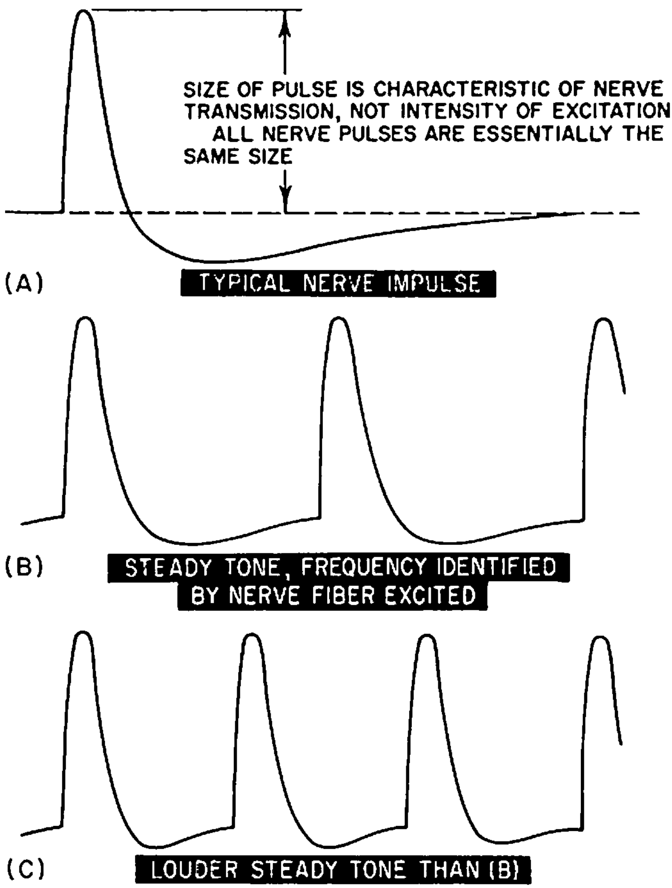
One point however, is essential here in helping to decide how important are different properties of reproduced sound. Because so many nerve fibers are used to cover the audio frequency range, human hearing is very sensitive to difference in *pitch,* or frequency. A small fraction of a musical semitone change is easily detected. On the other hand, intensity or loudness registers by the nerve pulse rate along the fibers. This means that only by critical listening can a change of intensity of 10% or even 20% be heard.

Our experience has come to associate certain groupings with which we are familiar with the particular sounds we recognize. This is really part of the action of our memory. We remember these sounds by patterns they produce and recognize them when we hear them again. This is the *conditioning* of our hearing faculty.

Realistic Sound

Without having heard similar sounds previously, a new kind of sound means nothing whatever to us and sounds completely strange. Sometimes strange sounds may bother us, while at other times we

Fig. 8. Pulse rate along an individual fiber of the auditory nerve conveys the *intensity* of the sound; the identity of the fiber tells the frequency.



may ignore them. This depends on whether we happen to be interested in something else at the same time.

For example, in the early days of radio and phonograph reproduction, distortion was enormous by modern standards. But at that stage we were only interested in hearing the program, and consequently all the strange sounds caused by distortion went unnoticed. Our attention was focused on the program material and our hearing faculty successfully ignored everything else. We even commented that the reproduction was life-like.

The hearing faculty however, has to contend with every sound the ear picks up — at a high enough level to be audible — whether we are conscious of it or not. So these strange sounds caused by distortion exercised the ear, though we were not aware of it. All we felt was a sense of relief when we removed the headphones and just talked to friends in the room. Later on we came to the conclusion that the reproduction was not as realistic as we thought it was.

Further attention and familiarity helped us to recognize the distortion components that should not have been there, and the fact that they marred the clarity of the reproduction. This general awakening is illustrated by the fact that some years ago it was "proved" at a demonstration that no one could hear less than 5% distortion. More recent tests have found that more than two-thirds of the people present could hear 5%, and one-third could detect 1%.

A little thought about the things we have discussed in this chapter will show how extraordinarily flexible the interpretive faculty of the brain really is. The effort of listening to our friend in a crowded room where many conversations are going on at once, makes our brain subconsciously turn down the volume on the general chatter.

The motor mechanic checking a fault in the engine, unconsciously turns down the volume on everything except the particular sound he is listening for.

When you pick up the sound of that aircraft overhead you may be near a highway on which there is considerable motor traffic noise that you can't fail to hear. The frequency components of the motor traffic undoubtedly overlap the frequencies radiated by the airplane. Yet somehow your hearing faculty ignores the traffic noise for a moment and concentrates on the particular sound of the airplane to determine its direction.

In any orchestral program different sections of the orchestra play different musical scores to make up the complete orchestration as the composer wished it to be heard. Listening, and as you become more interested in the music, you find yourself concentrating on the score as played by one or other section of the orchestra – the first and second strings, the wind instruments, even the drums. You will find you can hear quite distinctly the separate parts that each plays, although they will be made up of frequencies that overlap one another. This is because your hearing faculty has become use to grouping together the impulses coming from each particular instrument or group of instruments. So you can concentrate your attention on the first or second violins, for example.

Being able to do this however, depends upon the clarity with which the sounds are heard. If the musical components from the different instruments become all mixed up and distorted, so that the tones coming from each cannot be separated properly by the hearing faculty in the brain, then some of our appreciation for individual parts of the score is going to be marred. We may not even be able to separate the parts played by the different instruments as we should.

This is the reason for present day stress on reducing distortion in high fidelity equipment. Having brought it down to the lowest possible level however, something is left wanting and this seems to be a sense of realism, or presence, due to the ability of our two ears to distinguish between original sound and reverberation, and between sounds coming from different directions. When all the recorded sound goes onto one channel and is reproduced from a single loudspeaker, our hearing faculty is deprived of some of the distinguishing means present in live sounds or original sound.

Not that the sound we listen to is two-dimensional. The sounds from the loudspeaker still come out and fill the room, but we do not get the same kind of sound as would have come from the original orchestra or whatever program we are listening to. This is what led to the introduction of the various forms of stereophonic sound. But before we get to that, we need to understand some other things. In this chapter we have discussed the faculty of hearing and the part binaural listening plays in contributing to it. In the next chapter we shall straighten out a few things about the nature of sound itself and then we can see just what stereophonic sound really is.

Chapter 2

STEREOPHONIC SOUND

Before we can get an appreciation of what constitutes stereophonic sound we need to understand what sound waves themselves are. Sound waves travel in air because air is what the text books call "an elastic medium". This can be seen by using a common bicycle pump and pressing the handle in so that the piston or plunger moves down the barrel of the pump, holding one finger over the outlet so no air escapes (Fig. 9).

If the piston or plunger moves halfway down the barrel of the pump then the air, which originally occupied the whole barrel, is now compressed into just half the barrel. This can only happen because the air is compressible. As a result of this compression, the pressure of the air is just twice what it was before the plunger was pushed down.

How Sound Waves Form

This excess pressure can be felt on the finger holding the air in. The original air pressure could not be felt because it is the same as the air pressure all around us, called atmospheric air pressure (approximately 15 pounds on every square inch of every object in contact with the air). Pushing the plunger halfway down the barrel of the pump increases the pressure inside to about 30 pounds on every square inch and the pressure you can feel on your finger is the excess 15 pounds per square inch on the inside as compared with the outside.

Sound waves do not compress the air to anything like the extent your bicycle pump does, when you can feel the pressure change. In fact, an average sound wave only compresses the air by one millionth of its static pressure of around 15 pounds per square inch. A loud sound will produce a compression many times this, while a soft sound, one just audible, will only produce a pressure very much less even than one millionth of the steady pressure.

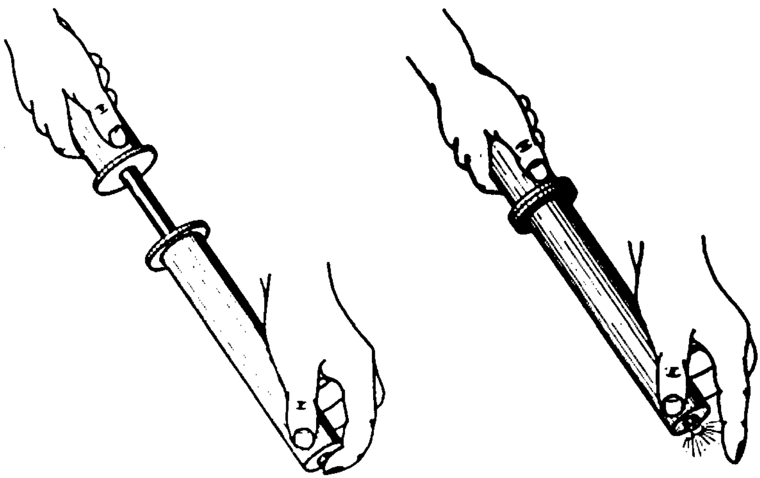
The air has to be moved to be compressed. In the case of the bicycle pump the plunger has to be pushed in from one end to compress the air in the other end. So movement of some air, without moving the air adjoining it, will create a change of pressure.

The opposite is also true. Difference in pressure creates air movement. If you release your finger, some air rushes out of the bicycle pump. These two facts combine to produce traveling sound waves.

Suppose a sound wave starts with a compression of air, one millionth above the normal air pressure at this point. This will start air movement away from the compression. Next,the movement away from the compressed area starts a compression in the area adjoining, because the next layer of air does not move immediately. So this layer gets compressed. This compression in turn starts the next layer of air moving and so on. The result is a wave of compression moving forward as shown in Fig. 10.

The movement of a sound wave is very similar in some respects to the movement of a wave on the surface of water. Compared to

Fig. 9. A bicycle pump can demonstrate the properties of air that make it transmit sound — compressibility and weight in motion.



AIR IS COMPRESSED

REMOVING FINGER RELEASES COMPRESSED AIR

air, water is practically incompressible. Consequently changes in pressure, due to movement of the water, do not result in the water being compressed into an appreciably smaller space, but cause the water to be lifted up at this point, so it has a greater "head". This produces the crest of a wave.

A wave crest will start water moving away from the crest so as to restore the level surface. This movement of water builds up a crest in an adjoining space and the result is a wave that travels along the surface of the water, very similar to a sound wave passing through air.

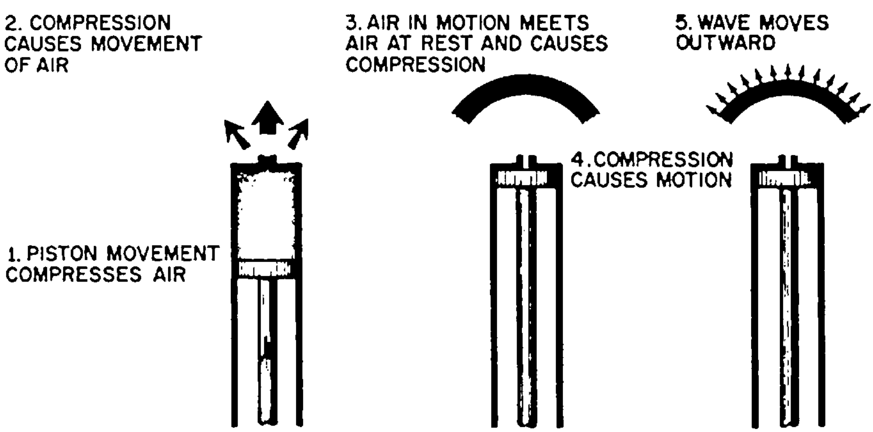
The main difference is that the wave appears only on the *surface* of the water as an advancing line, while a sound wave in air is two dimensional or has *area* and travels in the third dimension. This difference is shown in Fig. 11.

All the kinds of sound that our ears receive and pass on to the brain as audible "intelligence" are carried through the air by means of sound waves of this kind.

This fact is the first step toward an understanding of the stereophonic nature of all sound. When we listen to someone talking in the same room, or to a musical program in a relatively small studio, we hear the sound so closely after seeing the person speak or the orchestra perform that we naturally think of sound as being instantaneous and not taking any time at all to travel. The important thing to realize here is that sound is not instantaneous, but takes time to travel *however short the distance.*

If sound were instantaneous then reproduction would be a much

Fig. 10. The succession of events that cause a pulse of air to form a traveling sound wave.



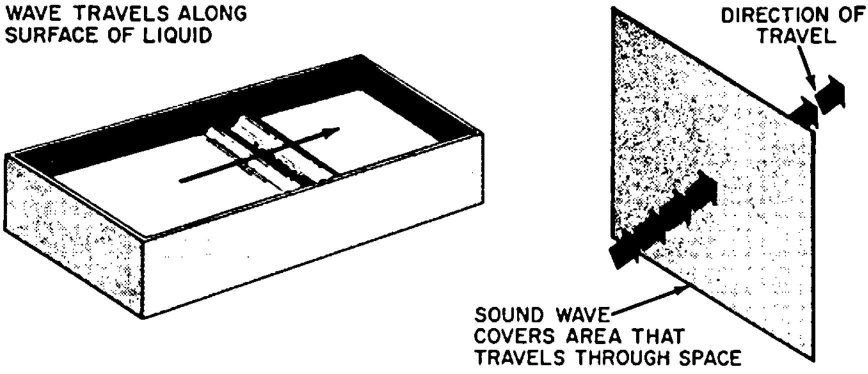


Fig. 11. A wave on the surface of water is a traveling *line*, but a sound wave in air is a traveling *area* (although the air itself does not travel).

simpler problem. All we would have to do is reproduce the same sound pressures in the listening room that existed around the microphone during recording. But the fact that sound waves travel means we need to produce an identical *movement* of the waves as well as identical pressure variations in the waves. This seriously complicates matters, but at least we can try.

Different Kinds of Sound Source

To start with, different sources of sound radiate the original wave in different fashion. Take first of all, the piano. The source of sound is really the string, which is struck by the hammer when the key is depressed. But the string itself does not radiate much sound. In fact a piano without a soundboard is almost silent. It requires electronic reproduction to make it audible. Such pianos have been constructed, but most pianos in everyday use employ a soundboard.

The vibrating strings make the soundboard vibrate by transmitting the vibrations through a bridge over which the string passes to the soundboard. Then the whole soundboard vibrates at the same frequency as the string and sets the air in motion.

Have you ever noticed that you can hear a distinct difference between a grand piano and an upright piano? This is not just because the grand piano uses overstrung strings. There are also overstrung uprights. But these do not give quite the same sound as a grand piano. The principal reason is that the soundboard of the two types of piano is in different positions.

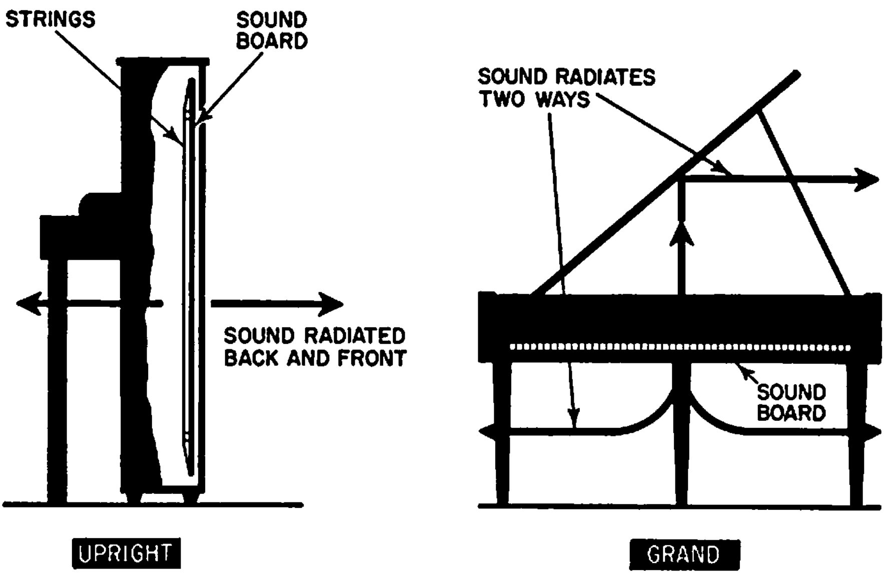
An upright piano radiates sound from back and front, while a grand piano radiates sound from top and bottom, the sound from the upper side usually being reflected by the lid of the piano, propped at a suitable angle. This difference is illustrated in Fig. 12. Because a trained hearing faculty can detect the subtle difference between the two kinds of wave radiated, we are able to tell the difference between the sound of a grand and an upright piano.

Another practical experiment that demonstrates the principle of sound radiation, and one you will know very well, uses the ordinary tuning fork. If you do not have an ordinary tuning fork used by musicians for tuning instruments, you can try a very similar experiment with the kind of fork used for eating.

If you twang the prongs of the fork to set them in vibration, holding the fork up in the air, very little sound is heard. But if you now place the base of the fork on a table, a clear tone becomes audible immediately (Fig. 13). You will notice that two things happen when you place the heel of the fork in contact with the table, or some other large surface: the sound becomes much louder and at the same time it drops pitch by one octave.

The vibrating prongs of the fork do produce some sound, due to alternate compression and lack of compression of the air adjacent to the prongs. But there are two compressions for every complete

Fig. 12. An experienced listener can tell the difference between an upright and a grand piano because of the different way each radiates sound.



cycle of movement of the prongs, one when the prongs are farthest apart and one when they are nearest together. However, this vibration of the prongs transmits a frequency corresponding to once per cycle of movement into the base or heel of the fork, and this transmits a similar vibration to the table or other body to which the heel is pressed.

This vibration is just half the frequency of the compression

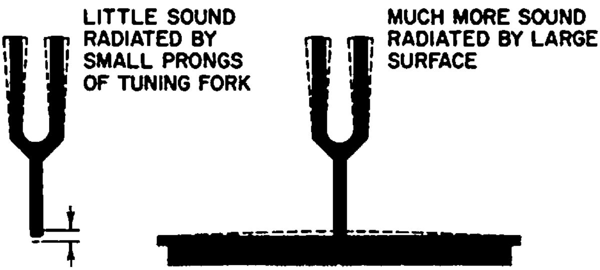


Fig. 13. A tuning fork radiates very little sound until its heel is brought in contact with a large surface, which does the sound radiating.

radiated by the prongs of the fork directly into the air, which is why we notice the drop in pitch when the heel is pressed against the table. The first radiation is a very small one due to the prongs threshing in the air without moving it much. The second one, which has a much clearer tone, as well as a louder one, is due to the rhythmic vibration transmitted to the large surface that moves a correspondingly larger volume of air.

This illustrates why stringed musical instruments need some kind of radiating surface to produce pleasant audible sound. The violin for example, transmits the vibration from the string through the bridge to the belly of the instrument. This is further carried by the sound post to the back of the instrument. Thus the body of the instrument vibrates in quite a complex fashion producing compression and rarefaction of air that is in contact with the outer surface, and also inside it, producing a radiation from the F holes of the instrument (Fig. 14).

The trumpet and other wind instruments use a different method of radiating sound. Here a pipe is made to resonate a column of air. The sound wave is produced directly instead of from a vibrating element like a string. The compressibility of the air and the motion at the mouth of the trumpet are used directly to produce a resonant effect that sets the frequency radiated (Fig. 15). This enables a greater quantity of air to be moved in a smaller area, because of the fact that the air itself is moved instead of just a small vibrating object like a string or the surface of the instrument.

Still another family of musical instruments utilizes both forms of vibration in combination. A marimba or xylophone, is an example of this (Fig. 16). Here the initial tone is produced by a vibrating piece of wood, struck by a hammer. This tone is not very loud by itself, but is amplified by the use of a resonant column of air, produced by a pipe hung underneath the vibrating wood.

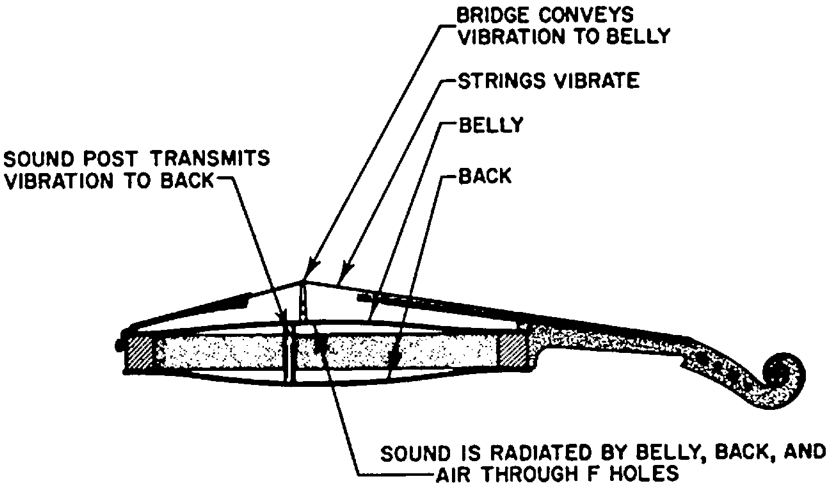


Fig. 14. A violin is a very complicated form of sound radiator.

When the wood is first struck the column of air is not vibrating at all, but the vibration of the wood gradually sets the air in vibration at the same frequency. The natural vibration of the wood produces a threshing action in the air, very much like the prongs of a fork or tuning fork. But gradually the pipe builds up the pure tones of the true frequency started by the wood.

This accounts for the peculiar characteristic sound of the marimba, xylophone and similar instruments. The first part of the sound is quite high pitched and characteristic of a piece of wood being struck by a hammer. The later, follow-through tone of the instrument has a pure sound, due to the tone sustained by the resonant air column in the tube.

The big violin or string bass (also the cello) uses still another means of transmitting sound to be heard. This time the instrument is rather small to transmit the very large sound waves for which it is responsible.

The lowest frequency radiated by a string bass is in the region of 41 cycles or vibrations per second. As sound waves in air travel at an approximate speed of 1086 feet per second, this means that the beginning of the first wave will be 1086 feet away by the time the end of 41st wave gets started. So each wave will be 1086/41 feet in length, or a little over 25 feet. A violin with dimensions this big would be rather awkward to play!

Consequently the instrument is fitted with a peg at its bottom

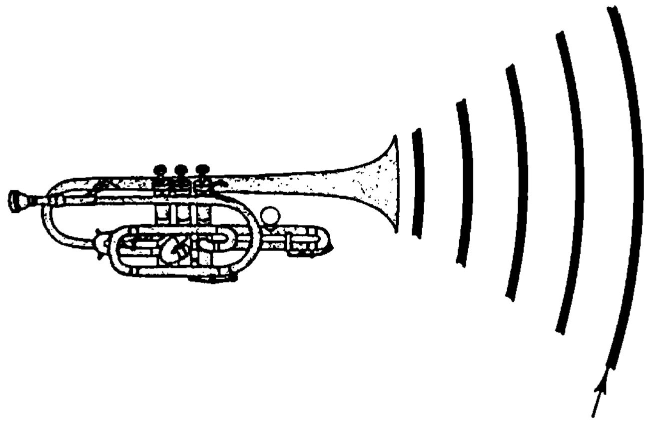


Fig. 15. A trumpet produces a direct form of sound wave radiation from its mouth.

end, by which it rests on the floor. The vibration from the string is transmitted by the body of the instrument to this peg, which in turn transmits the very low frequency vibration to the floor where it rests. As any bass player will tell you, it makes a great difference to the sound of his instrument, whether he rests the peg on a wooden floor or a concrete floor! On the latter the instrument is practically silent, while a good wooden floor will produce the pleasing tone for which the instrument is known.

As well as radiating the sound they make in different ways, there are ways the sound grows and dies away that are characteristic of different types of instrument or the manner in which they are played (Fig. 17).

This has been just a brief review of how some musical instruments produce and radiate their tone. It is by no means exhaustive and there are many other kinds of sound to which we listen, such as the human voice, which is produced by means of vibrating

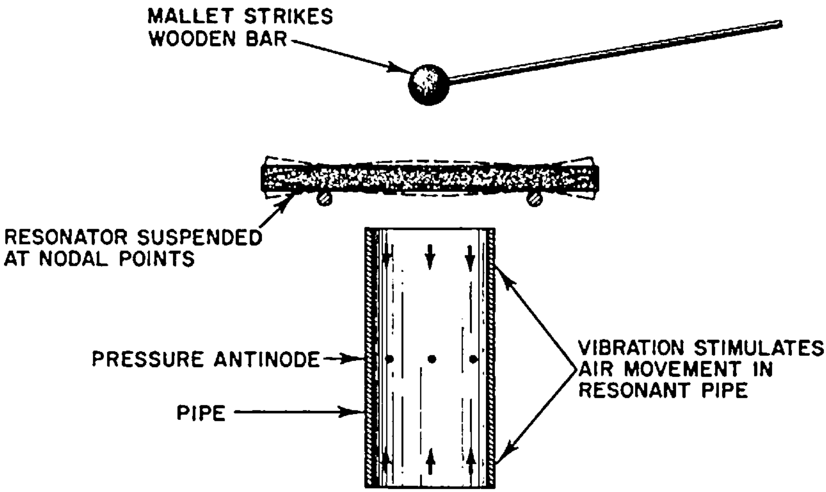


Fig. 16. A marimba or xylophone has a combined sound "generator" — a wood resonator with an air-column resonator that "follows" it.

vocal cords whose sound is further modified by the shape of the cavity we form with our months. This enables us to form the different vowel sounds, while the formation of the "release", such as teeth, tongue, lips, etc., enables us to form the consonant sounds. All this produces a very complicated pattern of sound waves which is radiated from the mouth of one person to the ear of another, and gets interpreted eventually into sentences conveying intelligence which we call speech.

Background Sound

But the making of original sound by all these different sound sources is only part of the sound experience that we pick up when listening to everyday sound. Regardless of what sounds you are listening to, you are also able to tell where you are hearing it, merely by the sound; whether it is out in the open air, in a field, in a city street, in a room, in an auditorium, or in a corridor. Each of these different places adds a characteristic to the original sound which enables you to tell where you happen to be listening.

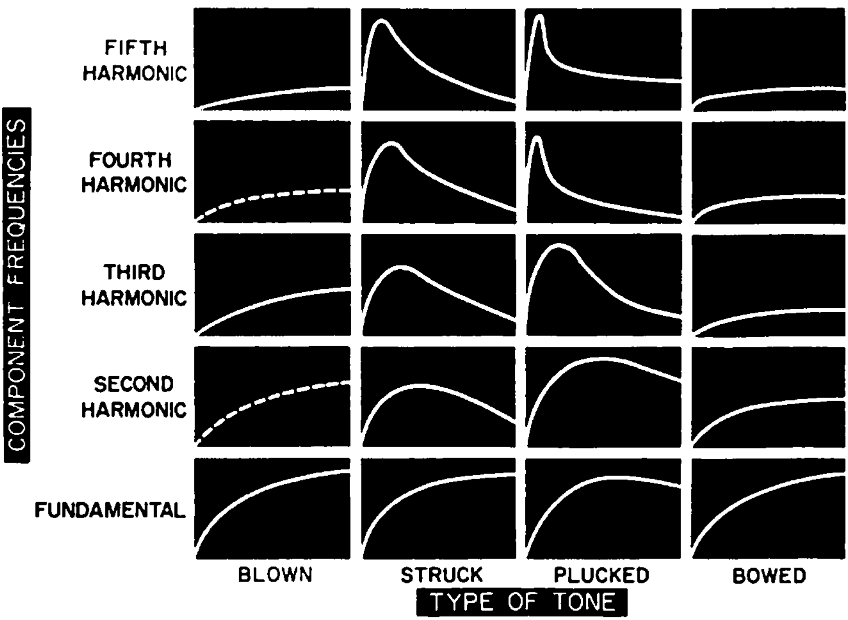
Even in your own livingroom it makes a big difference to sounds you hear there, whether you have wall-to-wall carpeting and furniture in the room, or whether the room is empty, as it may be when you first move in or move out of your home.

These differences are due to the effect known as reverberation. This rather technical word means practically the same as echo. The fact that sound travels and is reflected from surfaces with which it comes in contact produces secondary sounds to the original. An echo, however, is usually understood to mean a kind of reverberation where the reflected sound can be noticed as a separate entity.

When you hear your sentences come back to you from an echo rock it sounds as if another person is talking back to you, because the time taken for the sound to travel from you to the rock and back again enables you to be several syllables ahead of the reflected sound. But in most instances reverberation is of such a short period, the time difference between the original sound and the reverberant or echo sound is so short that you cannot hear them separately, not consciously at any rate.

However, the presence of the additional sound does make it sound different to your ear and you are aware of this difference as part of the overall sound. Actually, reverberation in most places is quite unlike the simple echo effect that you get with an echo rock. It is not just a single reflection, but a continued series of reflections. If we use the ripples on water as an illustration of

Fig. 17. Different ways an instrument is played produce different growth patterns for the frequencies it produces.



how a sound wave behaves we can see what happens by vibrating the water at a certain point, representing a sound source, and watching what happens.

First the waves travel outward from the original point of vibration. As they hit the boundaries of the tank, reflected waves are started back inwards. Eventually the whole tank is full of a mass of "standing waves" — waves that do not appear to travel, but cause the water to bob up and down (Fig. 18).

The same kind of thing happens in any enclosed space, whether it is an auditorium, a room, a corridor, or a street lined with brick walls. When you start vibrating the surface of the water, or start

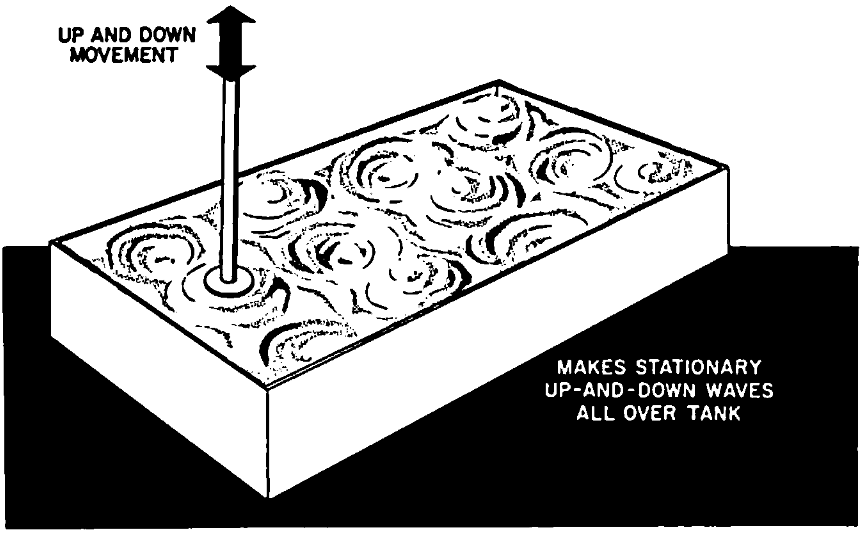


Fig. 18. Standing waves of the kind that make up room reverberation, can be illustrated by agitating water in a tank.

a certain tone in such a location, the waves take time to build up to this condition after the first traveling wave starts. Then, when the sound stops, there is a vibrating system of standing waves going on that again takes time to die away.

This build up and decay of the standing wave pattern that "follows through" from the original sound, is what gives the character to the reverberation. The different shape of the room or enclosed space, as well as the difference in size, adds a peculiarity to the way these standing waves build up and die down and this is what enables us to tell the difference in the kind of space where we are listening to sound.

The Importance of Transients

Now we can see that there are two distinct parts to the complicated pattern of waves to which we listen, that enable us to interpret the sounds we hear (Fig. 19).

There is the part consisting of initial sound waves, transmitted every time a sound starts up, like the waves that radiate from the original vibration in the tank. This is the first information we have of each component of a sound we listen to.

Then, following every tone in a sound, there is a tendency to build up a standing wave of some sort, which again decays after

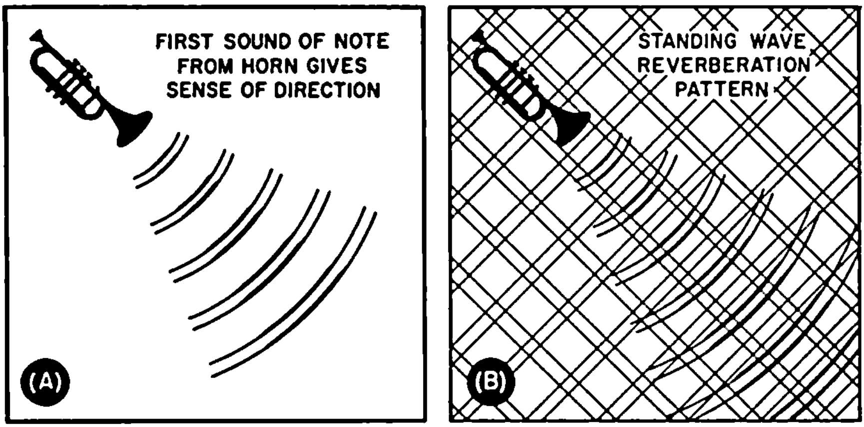


Fig. 19. Our hearing faculty subconsciously differentiates between the direct wave at the beginning of a tone (a), and the "follow-through" with its accompanying reverberation.

the sound has ceased. This is more of a nondescript type of thing — a vague pattern that builds up without a definite sense of traveling as the original waves do, but which does have a definite characteristic of its own.

The interpreting faculty of the brain that listens to all the waves received by the two ears, readily separates these two effects. This is why we are easily able to listen for the characteristic tones of different instruments, like the violin, wind instruments, and so on, in an orchestra, and separate them. It is because, when we listen in this way, we tend to ignore the effect of the follow-through reverberation. We just listen for the transient tones or the original sounds that reach our ear first, and analyse the sound pattern or component frequencies and pulse sequence they generate to recognize the kind of instrument.

Here we begin to see the importance of transients in sound waves for recognition purposes, both to recognize what kind of sound we are listening to, that is the kind of instrument it comes from, or what kind of object makes the sound, and also to recognize where it comes from, the sense of location, by binaural perception.

This can be particularly noticed from observations of sound from different objects. Sounds that have sharp, distinct transients, like a clock ticking, the striking of the musical triangle, any kind of instrument that is struck, such as a xylophone, or plucked such as a banjo or guitar, or the sharp, edgy tones of a person's voice calling our name; each of these sounds give us a keen sense of direction to locate the source of sound.

On the other hand, sounds that do not have such a sharp, distinct initial transient, such as an open diapason organ pipe, or the whistle or hum referred to in the previous chapter, obscure the sense of direction, because a standing wave is built up by these tones before an initial transient has an opportunity to convey a sense of direction or location.

This definiteness and indefiniteness about different kinds of sound source, depending upon the degree of transient effect they produce, is also noticeable to some extent in reverberation — the characteristics of reflected sound. Some forms of reverberation, such as the echo rock or a building with a large flat wall, or even a curved wall, produce discrete reverberant components that produce a definite echo effect. The original transient pattern of the sound wave is reproduced by the echo or reverberation coming back from the surface.

On the other hand, in a building where the wall surfaces are broken up and where there is no distinct large surface to produce a separate echo effect, they gradually build up standing waves and again gradually let them decay, taking perhaps a second for the build-up and another second for the die-away. In this case the reverberation does not have the distinct characteristic of the other type.

With the type that gives a distinct echo, the reverberation itself has a character, both as to kind of source and direction, similar to the original sound. This enables the direction of the reflecting wall surface to be detected in the same way as the direction of the original sound. But the more common sort of reverberation, which is quite indefinite in its character, builds up a kind of follow-through sound which does not have this definiteness. Consequently our hearing faculty tends rather to ignore this reverberation — to reject its sound as being not part of the thing we really listen to — although we are conscious of it.

Our hearing faculty in effect separates the original transient from the sound source — the orchestra, the conversation or whatever we are listening to — from the general follow-through tones that the reverberation adds, which have no particular sense of direction but are like the standing wave patterns built up in the water tank that grow and die away as a whole rather than as a traveling wave.

To avoid possible misunderstanding or confusion, I should explain that the word "transient" is used here a little differently from the way an audio engineer uses it. In this book "transient" will be used to describe an acoustic, or sound wave, where its *traveling* property is clearly distinguished as opposed to *standing* or *stationary* waves, where the whole area is filled with fluctuating pressures without any appreciable sense of "movement" or source.

It is to try to convey some of these differences in the character of the *original* sound more effectively, that stereophonic sound is introduced.

Early Progress Toward Fidelity

In the development of audio amplification from the stage where it was just a part of a radio receiver that amplified the sound frequencies after their detection, or a simple electrical amplifier for a phonograph to replace the old acoustic horns, up toward what is now known as "high fidelity", the early work consisted of endeavoring to make sure that all the frequencies making up the sound wave — from 20 cycles up to 16,000 cycles — were amplified uniformly, and that no distortion occurred that would give rise to any foreign frequencies not in the original program.

At first it was not difficult to make improvements in this direction. Then, as the distortion, both of frequency response and waveform, was reduced to lower and lower levels, it seemed we *should* have reached the point where deviation was no longer audible and the reproduction would sound perfect. However, the critical listening faculty of hi-fi addicts was not satisfied. The something, as I said in Chapter 1, that would have supplied realism, was still missing from this reproduction.

Careful listening showed it was important what *kind* of loudspeaker you used as well as *how flat was its frequency response* and how well it avoided any kind of distortion. Not only does the loudspeaker have to radiate all the frequencies present in the original sound at uniform and proportional intensity to the original, and without any distortion, but it must also radiate waves that have a similar radiation pattern to the original if a reasonable sense of credibility is to be achieved, that is to say, if the reproduced sound is to be believably like the original.

For example, a relatively small, single-unit loudspeaker gives quite a realistic reproduction of speech — the spoken word — because all the sound comes from a relatively small unit, not much bigger than the human mouth. But such a loudspeaker gives an ineffective presentation of an orchestra. One of the large type loudspeakers that uses a number of units to cover different sections of the frequency response, and distributes the apparent source of sound over quite a large area, gives a much more convincing reproduction of an orchestra.

Reproduction of a grand piano seems to sound most realistic from a bass-reflex speaker in which sound comes from the front of the diaphragm at one point, and also passes through the cabinet from the rear to a port, making another component of sound coming from another point. As this closely resembles the way a grand piano radiates its sound, this kind of loudspeaker is particularly realistic for this kind of sound. It tends to be somewhat less realistic for the tuba, for which purpose one of the horn-type radiators seems to give the more realistic reproduction.

This is also true for instruments like the trumpet, cornet, trombone etc., where the sound comes from a horn mouth which has the property of projecting, particularly the high frequencies, almost in a straight line from the mouth. A loudspeaker that uses a horn-type tweeter gives the most realistic reproduction for these. A loudspeaker using a small-diaphragm cone unit seems to sound quite woolly when reproducing this kind of instrument, although its frequency response may measure almost identical.

The best kind of speaker to use to achieve realistic reproduction, has been found to depend, not only on the kind of program material you want to reproduce, but also on the listening room. In a fairly small livingroom, reproduction of an orchestra may sound quite reasonable from a fairly small loudspeaker. Putting the same loudspeaker, reproducing the same orchestra, into a larger room loses the realism, because the orchestra now seems to be coming from a single *point* source of sound. This is where the larger type of loudspeaker is particularly necessary to get realism in orchestra reproduction.

On the other hand, the disadvantage of the larger speaker, that it spreads the vocal reproduction and makes a person's voice seem to be unnaturally scattered about, is often minimized in a large room, because the listener will be sitting at a considerable distance from the loudspeaker. This gives the sound from the different units an opportunity to merge together by the time it reaches the listener's ears. In a smaller room there would not be sufficient space for this to happen.

The Toneless Loudspeaker

In earlier days an ideal had been established that the reproducer system should do precisely that, i.e., *reproduce,* without adding any characteristics of its own. As far as frequency response and distortion is concerned, this ideal can be approached quite closely, by minimizing the deviation in the amplification of different frequencies and reducing the amount of distortion produced, to a point where the deviation is certainly not audible. But it is definitely not possible to have the loudspeaker radiate the sound in exactly the same manner as the original sound was radiated.

Following the general idea of not adding any character of its own, the first thought about the ideal way a loudspeaker should distribute was that all the frequencies should be distributed uniformly in all directions. At low and medium frequencies this is no problem. The loudspeaker naturally distributes them this way. The sound waves are fairly large and it doesn't take many of them (at the low frequencies only a fraction of a wave) to fill the room. Consequently there can't be very uneven distribution of the sound at the low and middle frequencies.

But at the higher frequencies the waves are very short compared to the dimensions of the average listening room. This means that they can travel around in narrow "bunches". The result is that sound may be projected in a narrow beam from the loudspeaker so that one position will get all the high frequencies and other positions apparently don't get any — by direct radiation from the loudspeaker, that is. However, the high frequencies can still more or less fill the room by bouncing off the various wall surfaces in a form of reverberation.

This seemed like a defective situation, so a variety of loudspeakers were designed with the object of uniformly distributing the high frequencies. Acoustic lenses and various other kinds of diffusers have been utilized to ensure that the higher frequencies were spread around more so that all parts of the room would receive an equal proportion. At first this sounds like a good idea, but further thought shows that it cannot give a realistic impression of the different *kinds of sound source* discussed earlier.

At this stage in the discussion, probably the biggest case *for* stereophonic sound comes from the reproduction of orchestral music. Although the large loudspeaker with a number of units does give the impression of a bigger sound source and thus is better than the single-unit type, it still lacks the spread or broadness the natural orchestra can give. The large loudspeaker gives the low frequencies at one point, the middle frequencies at another and the high frequencies from still a different position in the orchestra.

The sound from the violin covers or requires quite a *range* of frequencies which will be delivered, in the loudspeaker, from the middle and high-frequency units. The wind instruments likewise use a similar range of frequency. So the large loudspeaker unit virtually takes the lower frequency components of both the strings and the wind instruments and delivers them at one point, while the higher frequencies of these two groups gets delivered at another point. The distribution cannot be correct for the orchestra.

What we need is a loudspeaker system that will give us the sense of space present in an actual orchestra. This need was responsible for the idea of stereophonic reproduction. The original idea for stereophonic reproduction suggested a whole line of microphones arranged along one wall of the studio where the orchestra or other program is performed, and to correspond with it, a similar line of loudspeakers placed all along one wall of the listening room. Each loudspeaker relays the channel of sound picked up by the corresponding microphone (Fig. 20).

The expected result was that the exact form of the sound wave reaching the wall of microphones would be reproduced in the listening room. Experiments were conducted with this kind of system and considerable improvements were noticed. Then, to make the system economic so that people like us could buy one, experiments were conducted with less than the original number of microphones and loudspeakers. It was found that three microphones, feeding by three separate channels ultimately into three loudspeakers gave an optimum degree of realism — one that was not appreciably improved by adding more channels — while using only two channels gave a realism much better than obtained with just a single microphone and loudspeaker, but noticeably below the standard achieved by the three-channel system.

So three-channel was acclaimed as "true" stereophonic reproduction. This should enable us to sit in the listening room and tell exactly where the violins, the wind instruments, and various other sections of the orchestra are located in the studio.

But just a minute. Can you really tell, by listening alone in a concert hall, exactly where the different instruments are located in the orchestra? If you have ever tried from a position more than ten rows back, I am perfectly sure that you can't. You identify where

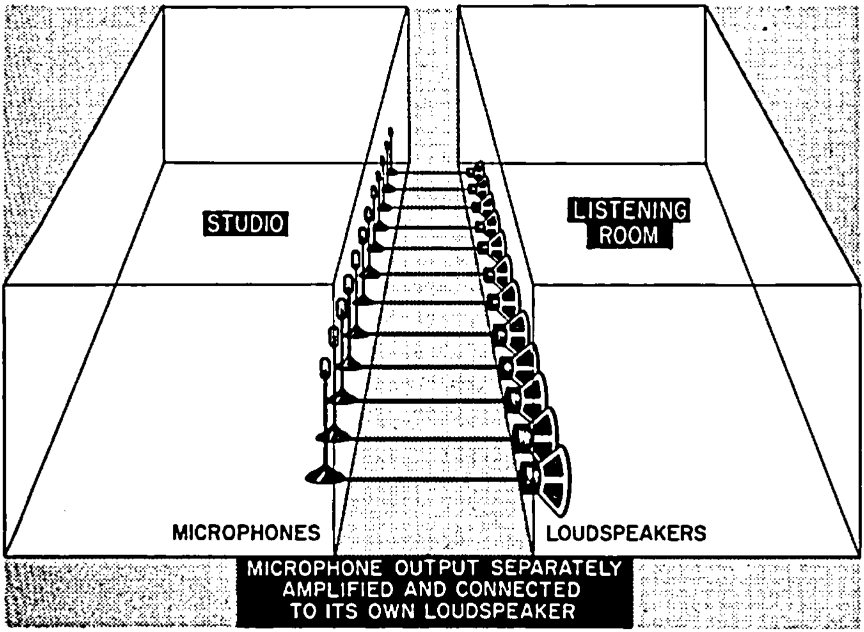


Fig. 20. The idealized stereophonic system shown here, endeavors to "remove" the double wall between studio and listening room by using enough microphones and loudspeakers.

the different sections of the orchestra are located by watching them perform, rather than by sense of your ears alone.

So assuming our newly acquired stereophonic reproduction *can* enable you to tell exactly where the different sections of the orchestra are located, it is doing something more than listening to a live reproduction can do. In other words, it is *better than live!* Can this be?

Stereophonic Theory

Meanwhile, the people who "like to have a good theoretical background for everything they do" set to work to present the *theory* of stereophonic reproduction, just to provide a satisfactory basis for setting up this three-channel system. All the previous work on sound analysis had used the basis or hypothesis that sound waves may be analyzed into component frequencies of from 20

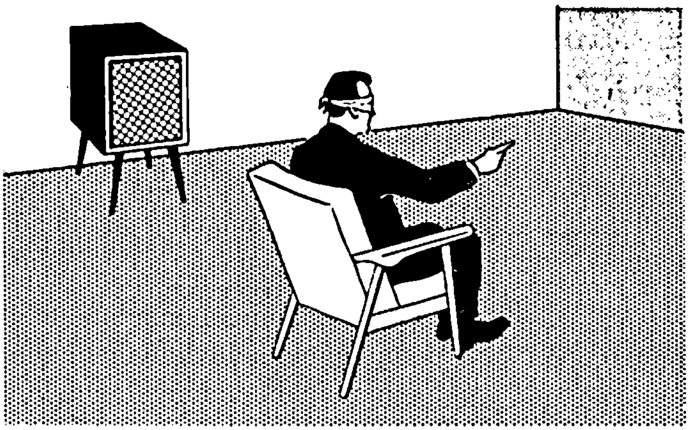


Fig. 21. Experimental determination of the human ability to identify direction of sounds at various frequencies.

cycles to 16,000 cycles (or some other limit). The theory ran that, if you treat each of these frequencies in the spectrum correctly, then the composite will be correctly handled.

From this basis arose the discussion as to how our binaural hearing perception can locate direction. The matter was investigated at different frequencies. Tests were made, using single tones to find how accurately direction could be identified, and what properties of the wave the brain "uses" for the purpose (Figs. 21 and 22).

At some frequencies the sense of direction achieved seemed to be due to the difference between the intensity of the sound at the two ears. At other frequencies it seemed to be due to the fact that the sound arrives at the further ear a little later, and thus what is called the phase difference, or the difference in time of arrival, was responsible for the sense of direction. Let's examine what this means in terms of a few typical audible frequencies.

First take a frequency about the middle of the range, say 1000 cycles. At this frequency, the wavelength of sound is about 13 inches, because it takes 1000 waves to fill up the distance traveled in one second, of 1086 feet. If the sound comes from directly in front of the listener, it will reach both ears at the same time and in the same intensity. If it comes from the left-hand side then it will arrive at the further ear a little more than half a wavelength, or period of vibration, later, and also the intensity will be considerably reduced because in passing the head a "hole" is made in the wavefront which has to be filled in by the wave spreading

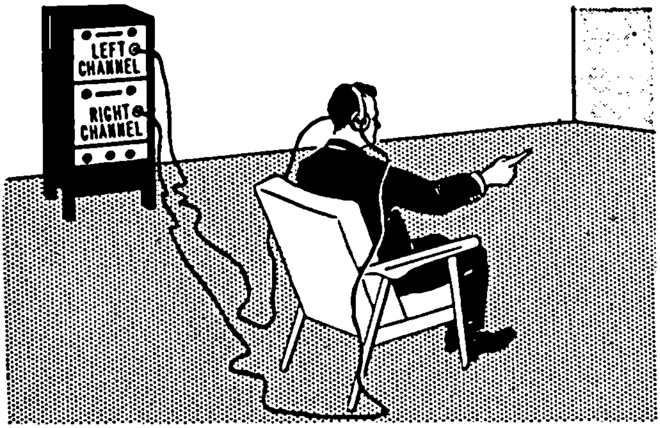


Fig. 22. Carrying the experiment of Fig. 21 a stage further, phase and intensity differences are synthesized in the headphones to find out what properties of the sound wave the hearing faculty uses to determine direction of source.

around the far side of the head (Fig. 23). This means the intensity of this part of the wave will be considerably reduced.

So it seems logical to assume, at this frequency, that the sense of direction could be due to difference in both intensity and time. Experiments, feeding a listener with controlled amounts of the same frequency by means of separate headphones (Fig. 22), prove that this is true.

At an extremely low frequency, say 100 cycles, the wavelength is about 11 feet. Consequently the hole made by the head in a wave is a very small fraction and the intensity reduction due to traveling around the head is quite small. There is still a time difference due to the fact that the wave has to travel from one side to the other. The wave is 11 feet long, and the distance from one ear to the other probably only about 8 inches, so the phase difference or fraction of a wave represented here is only about 1/18 of a full period.

This gave rise to the question whether the human hearing faculty can detect such a small difference in this low-frequency wave timing. Experiments seem to suggest that to a very small degree it can, but the accuracy of detection is very much poorer than at higher frequencies.

Going in the other direction, using frequencies between 5000 and 10,000 cycles, the wavelength gets to be in the region of an

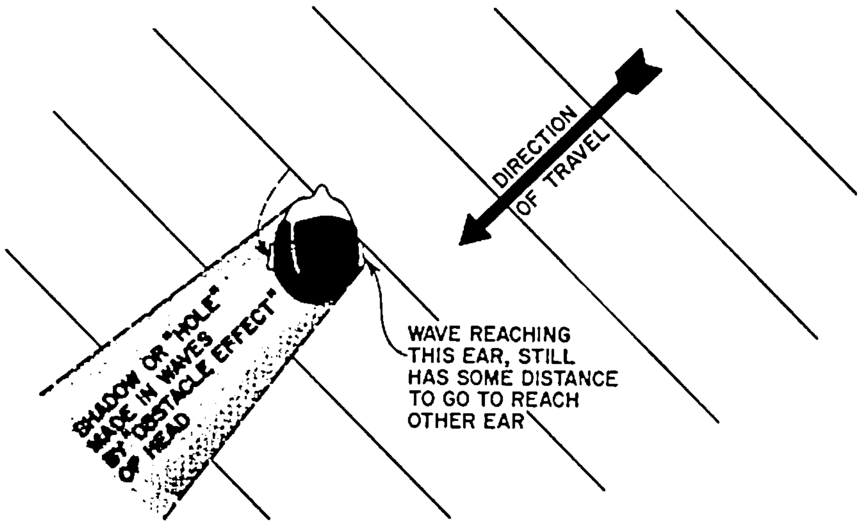


Fig. 23. In the region of 1000 cycles the arguments of the academic theory seem fairly plausible.

inch or two. This means the human head knocks out quite a sizable chunk of the wave arriving at one side, and consequently the spread of the wave around the head produces much more attenuation of the wave arriving at the other side (in the hole) than it did at lower frequencies. At the same time the phase difference may be *several* complete periods. It may well be that compression components of the wave arrive at both ears at exactly the same time, the difference being that one ear receives it one or two complete periods ahead of the other (Fig. 24).

From this, the obvious deduction was drawn that a sense of direction at high frequencies must almost entirely be dependent upon the attenuation, or reduction in intensity, from one ear to the other.

With continuous-tone testing, experimental results confirm our deductions and the direction-sense of the hearing faculty is far inferior to everyday listening experience. The suggestion has been made that the brain utilizes composite information drawn from an "analysis" at all the frequencies present in composite sound. In fact a considerable amount of arguing has gone on about different parts of this theory.

The one fact that often seems to have been overlooked is the presence of standing waves, whenever continuous tones are repro-

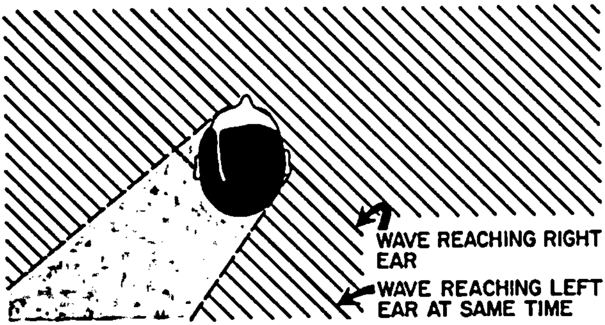


Fig. 24. At high frequencies "phase" becomes virtually meaningless, as there will be several complete waves between the two ears.

duced or radiated in a confined space such as a listening room. Tests can be (and have been) conducted in an "anechoic chamber" (a room where all the walls are so heavily padded that practically no reflection occurs at all). Under these conditions it is possible to obtain some verification of the theory just discussed.

But who listens in an anechoic chamber? If you have ever been in one, you will have quickly realized that it would be a most unpleasant place in which to listen to high-fidelity music. The complete absorption by all the walls gives the room an extremely dead or silent effect and reproduction in such a room seems entirely unrealistic. So for things to be at all natural, we must have the reproduction in a listening room with a certain amount of natural reverberation or reflection from the wall surfaces.

As soon as we do this, any test with continuous tones like that just described will produce standing waves, especially at frequencies above 100 or 200 cycles where there is room for at least three or four waves in the room at once. This means a pattern of standing waves will be set up, according to the size of the room. The waves will not be traveling around the room (according to the theory), but will be just like the standing waves in the water tank where the different points on the water merely bob up and down instead of the waves traveling. Correspondingly the air pressure at different

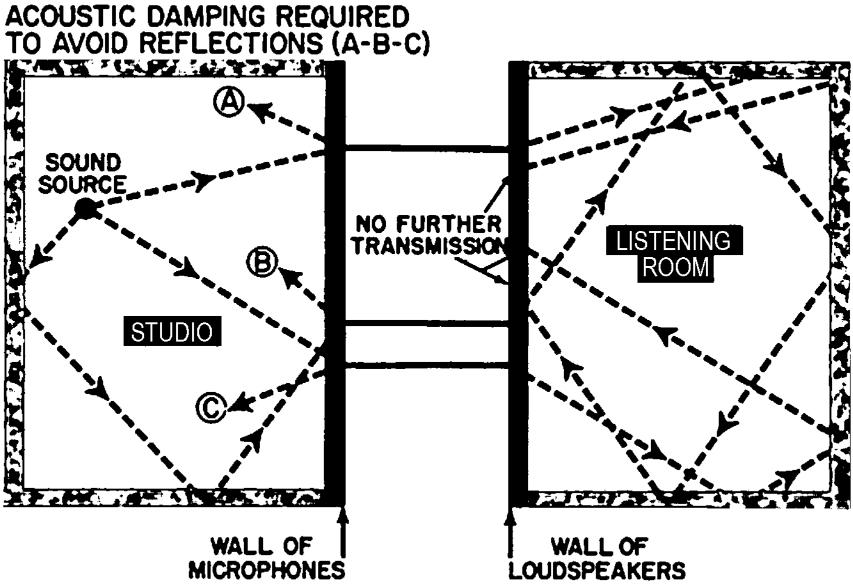


Fig. 25. A closer analysis of the "ideal" arrangement of Fig. 20 shows it cannot possibly achieve its theoretical objective, regardless of the number of microphones and loudspeakers used.

points in the room will be fluctuating without any apparent *traveling* wave effect.

It *is* possible to get some kind of directional sense from a standing wave, but only if you hold your head very still. As soon as you move your head, the relative intensity and phase at the two ears rapidly changes, consequently the apparent direction will seem to change very rapidly too.

So this method of making tests really tries to determine how we sense the direction of reverberation — a kind of sound in which we don't as a rule bother much about direction. The part we are really interested in is the direct sound, where the waves actually do appear to travel, and this is concentrated in the transients of the reproduced program.

How does this affect the realism achieved by the theoretical stereophonic system we discussed a little earlier — the three microphones feeding the three loudspeakers? The basic theory suggested that connecting each microphone with its corresponding loudspeaker, was equivalent to taking away the wall between the two rooms. The fact that the two rooms might be separated by a considerable distance, by radio transmission or time, making a recording and reproducing it, does not materially affect the situation. All the sound waves arriving at the wall occupied by microphones are accurately reproduced at the wall occupied by loudspeakers, by reproducing the sound pressures at the different points along it.

But notice that this theory assumes (although it was never so stated) that the sound waves travel *only* from the studio into the listening room, and that none of them go back (Fig. 25).

To achieve this idealized condition then, the walls where the

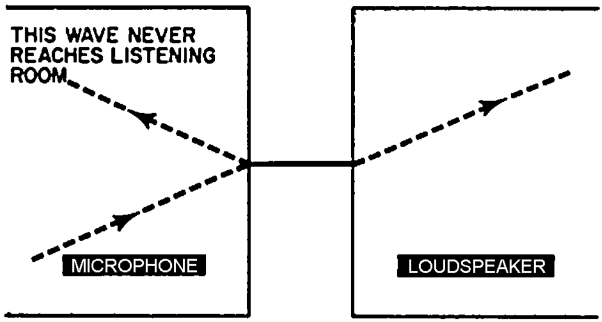


Fig. 26. Alternative to the use of non-reflecting walls behind the microphones (Fig. 25) and loudspeakers, reflecting walls can be used, but result is no nearer the theoretical ideal.

microphones are should be acoustically damped to prevent any reflection, and so should the wall where the loudspeakers are. Then reflected waves from the other three walls of the studio will also reach the microphones and be recorded or transmitted and given out, along with the original waves, at the loudspeakers. At this point the partial reverberation from the studio will be continued in the listening room and completed by reflection from the remaining three walls of the listening room.

This is one idealized possibility. But notice that this possibility assumes that the reverberation, as well as the original sound, can travel *only one way* through the intervening wall. It can reflect around the three walls of the studio, but once it passes through the fourth wall, it has no means of coming back. In any natural auditorium, studio, or listening room, it is possible for the sound to reflect from all the walls.

Maybe you should make the fourth wall of the studio, and also that of the listening room, reflecting. But this again doesn't fill the bill exactly, because now the wave reflected from the fourth wall of the studio doesn't get into the listening room at all (Fig. 26). To be more realistic the wave reflected from the fourth wall of the studio should be reproduced at the opposite wall of the listening room.

All this has been discussed, not to show that stereophonic reproduction is impossible, but that it does not achieve the proposed ideal, presented in the theory. The use of a stereophonic system does considerably improve the sense of realism given by the reproduction. But at this point I want to make quite clear the fact that such a system cannot *accurately* reproduce the original performance in the livingroom, whatever is done with it. It is always a compromise — and a much bigger compromise than is often stated in literature on the subject. The fact that two or three channels are used to record more information about the original program naturally means that we have a better chance of achieving a sense of realism in the reproduced sound.

But before we go on to consider a little more detail about stereophonic systems actually used, let's evaluate how important are the various facts we have discussed in this chapter.

The Big Help

First we showed how the part we are most interested in is the *direct sound.* The reverberant sound is necessary to get a sense of realism, but we do not pay such close attention to it. We do not analyze where the reverberation actually takes place to the extent that we pay attention to individual sounds coming from the original program. Our ears even have a natural tendency to ignore the reverberation although we are conscious it is there.

This fact is really the salvation of stereophonic sound, and indeed of any system of reproduction. Provided the reproducer system makes a good attempt at reproducing the sound transient waves, those coming from the original program source — the orchestra, instrumentalist, person speaking, or whatever it is — and provided the combined reverberation due to sound in the original recording studio, together with that added by reverberation from the reproduced sound, gives a realistic overall effect, we shall be reasonably happy that the reproduction sounds real.

The experimental results we discussed, while appearing to confirm, to an extent, the theory on which the experiments were conducted, only do so for a somewhat different reason. The most prominent sense of direction we can notice is not due to any sustained tones at all, but is due to transient sounds. Even the sound of an aircraft is a complex sound — a repetition of pulse-type sounds from the motors — containing a random mixture of the higher frequencies.

It is true the ear can give only a less critical sense of direction to the lower frequency tones. But this is not so much because of the long wavelengths involved, as the fact that, being a low frequency means they cannot *happen* relatively suddenly. Low frequency tones, by their very nature, have to take a considerable time to build up, and the time they take to build up is very much longer than the time taken for the sound to travel from one ear to the other.

However, a particular example will well illustrate the principles we have discussed. Fog signals used at sea invariably consist of some kind of horn that is mechanically blown to produce quite a low-frequency tone. The reason for using a low frequency is that it carries over a greater distance without absorption by the fog. Higher frequencies are more liable to attenuation with distance and might thus be obscured more quickly. But in the open space at sea, direction can very easily be recognized by listening to these pulses of quite low frequency.

Certainly the accuracy in sense of direction achieved under these conditions, is much closer than would be expected on the basis of the theory and experiments discussed earlier. This is because although the phase difference, measuring along the low frequency waves, is only a very small fraction, there is still a *time* difference in the arrival of the *start* of the wave, that the ear can utilize. The nerve fiber carrying information to the brain about this low frequency uses pulses just as sharp as those for high frequencies along other nerve fibers. So the brain can just as critically measure the time difference between these pulses, whether the frequency is low or high.

But on a continuous tone the brain loses the ability of measuring *time* difference between ears, because each ear transmits a continuous sequence of pulses not related in any way to the *phase of the audio wave,* but to its *intensity* at that ear. Thus our deduction that the sense of direction is based primarily on transients proves to be correct for low frequencies as well as higher ones (Fig. 27).

In reproduction for high fidelity in a livingroom, conditions are very different from those at sea. We are now enclosed in a limited space and the waves no longer travel freely past our ears because the room in which we listen can only contain, at the most, one or two waves at this frequency. A standing wave is built up immediately the frequency starts. So a foghorn reproduced over a loudspeaker in a livingroom will not give its listeners the same accurate sense of direction as the real foghorn. Possibly, if the sound is accompanied by some of the rushing of air through the horn, this hissing sound will give a sense of direction in the livingroom. But this is not the same component that helps the sailor to determine the direction when at sea.

The fact that the ear achieves a better sense of direction at the higher frequencies is because these are the frequencies of which transient sounds are more prominently constructed. The tick of a clock, or the "ting" of a xylophone being struck, is rich in the high-

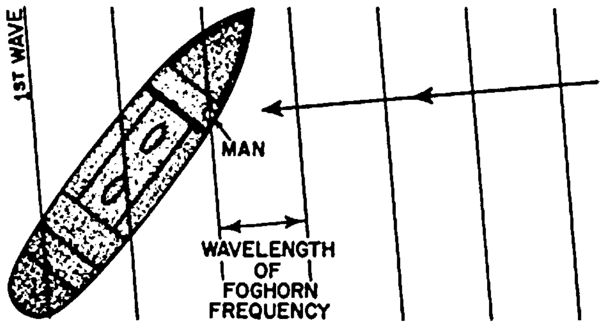


Fig. 27. Low frequencies, such as used by a foghorn at sea, can give very satisfactory directional identification, but they must be used in large, if not open, spaces.

frequency harmonics for a very short duration of time, and this makes the radiated sound extremely easy to locate.

Undoubtedly the intensity difference noted in the experiment is a contributing factor, but still more important is the *time* difference. This is proved by the fact that a person slightly deaf in one ear — so that normal intensity and time differences no longer apply when the impulses reach the brain — can still have a sense of direction in his hearing, not so very much inferior to that of a person with two equally good ears. The hearing faculty can make an adjustment for a permanent intensity difference and leave the brain still with the ability to determine direction on the basis of the time difference observed.

A person with one ear better than the other will probably notice he has difficulty in locating soft sounds whereas louder sounds are much easier to sense. This is because his stronger ear is stimulated much more intensely by the soft sounds, whereas his weak ear may not be hearing them appreciably, if at all. As soon as the sounds are stronger, the correction factor applied by the brain for intensity difference can do a much more efficient job, and the time comparison mechanism can more accurately determine the direction from which the sound comes.

Now we have some understanding of both sound and the hearing faculty, so we can discuss the systems available for use either in the home or in theaters, with a much better knowledge of what to expect them to do for us, and also with a better means of judging the prospect of each system in this regard. In these two chapters we have not discussed all the problems that will be encountered in making a stereophonic system — only those that are basic to our hearing faculty and the reproduction of sound in a room. Other problems relating to the systems themselves we shall discuss as they arise when we come to them.

The discussion in these chapters has shown that the "classic" theory presented to support either the binaural reproduction, using headphones, or the so-called stereophonic reproduction, using microphones with wider and identical spacing, does not fully explain the problems. Nor is either system able, even in theory, to achieve perfection of its objective when more closely examined. It is apparent that the important thing is the production of a satisfactory *illusion* of realism, though to do this we may have to use what some idealists have facetiously dubbed a "bistereonauralphonic" system, something which is a bit of each, or even nothing at all, according to "classic" theory.

As we progress, we shall analyze the problems encountered on the basis of the best prospect for producing the desired illusion, considering what is necessary for this purpose, rather than trying to get the nearest approach to some purely theoretical ideal, such as identical microphone/loudspeaker placement.

Chapter 3

STEREOPHONIC SYSTEMS FOR THE HOME

There are two ways to divide the different systems available or proposed for use as stereophonic systems. One of these concerns the difference between the program or information contained in the separate channels of the system. The other concerns the kind of material in which the program is conveyed, either recorded or transmitted. As there is no basic relationship between the two ways of dividing the systems, we will consider each separately.

First we will discuss the different ways of dividing the program or information between different channels. For this part, the discussion is concerned with how realistically the program can be presented in the livingroom. From the discussion in the preceding chapters we realize that the livingroom is fairly small, compared to many spaces in which sound may be recorded.

Specifically, the lower audible frequencies have wavelengths so large that the livingroom may only contain one wavelength, a fraction of a wavelength, or at the most just a few wavelengths. In any event, frequencies in this range do not have room to travel around as they would in free space. Standing waves are immediately built up because of the restricted size of the room relative to wavelength.

At the *very* low frequencies, where only a fraction of a wavelength can be contained in the room, the loudspeaker does not produce a complete wave at all, but virtually pumps air alternately in and out of the room at the desired frequency. So there is no stereophonic system that can do anything to give a sense of direction or discrimination to the extremely low frequencies. The sound coming from any number of loudspeakers will be approximately of uniform intensity and in phase. (If the loudspeakers are out of phase, cancellation will occur in the room, and diaphragms will move much farther than they should.) Consequently, with correct connection the effect must be cumulative, as it would be if a number of loudspeakers were connected to the same single-channel program.

It should be stressed that this remark applies only to the very low frequencies. At middle and higher frequencies intensity and phase differences (according to the theory usually presented) can be of importance. To be more precise, each channel will carry different information about the transient components of the sound in the middle and upper frequency ranges.

Basic Systems

In considering the potentialities of different systems and also examining the performance and the degree of realism they reproduce, we should keep in mind both the critical and the tolerant aspects of the human hearing faculty. It would be utterly fruitless, for example, to go to great expense to accurately reproduce all the different components of reverberation. This will add very little, if anything, to the realism of the presentation. On the other hand, it *is* worthwhile expending effort to get realism in the reproduction of the transient components that convey information equivalent to the original program material.

The reverberation part is not unimportant; it has to be there to achieve realism. But it is not important to achieve accurate precision in the reproduction of the reverberation.

For the home user economy is an important factor. The system has to be made at a cost that a reasonable number of people can afford to buy, or it is not a commercial proposition.

This applies not only to the reproducing equipment, but also to the recording medium used, where the program is recorded rather than received over radio channels. This we shall discuss in more detail when we come to consider the different media. For the moment we can think of the question of how well the channels are utilized – whether the number of channels we have is used to good advantage – because whatever kind of medium is used to carry the information, the cost of recordings is usually dependent on the number of channels required, and the time.

Closely associated with economy is compatibility. Can the stereophonic program play over a single channel and still give high fidelity? This question is related more closely to whether we use radio, disc or tape, and is discussed later.

1. Two-Channel Stereophonic

This system uses two microphones and two loudspeakers. The microphones may be spaced apart by the approximate distance of the human ear, in which case the proper technique for playback is to have the loudspeakers as far apart as possible, so that in a normal listening position, each ear hears *predominantly* the loudspeaker reproducing its channel. This way of using two-channel stereophonic is basically an adaptation of binaural reproduction. The recording is made in exactly the same way as the binaural recording, but the reproduction uses loudspeakers.

An alternative, and more generally used placement of the microphones for recording, puts them a foot or two (or maybe even more) apart, so the two channels receive the program with a bigger time and intensity difference than will occur at the spacing corresponding to the human head. This method is based upon the simplification of the stereophonic theory we discussed in the previous chapter.

Whichever method of recording is used, the program material recorded on the two channels is very similar. Listening to one or the other, the differences require critical listening to detect. But there are differences both of intensity and phase.

It should be noted here that the differences are not just in the same program material: it is not that the composite audio has its intensity and phase modified in the two channels; different components of the audio, such as violins, wood-wind, brass, etc., have varying combinations of intensity and phase.

Thus, if the violins for example are playing nearest one microphone while the drums are nearest the other, number 1 channel will carry the violins at higher intensity than the drums, and the phase of the drum reproduction will be just a little behind that on number 2 channel. Number 2 channel, on the other hand, will carry the drums at greater intensity than the violins and phase of the violins will be a short time behind that of number 1 channel.

Even so, whichever method of pickup is used, the time difference between the two channels is extremely small and the intensity is not usually very great either. Consequently you could listen to either one of the channels singly and scarcely note the difference. It is only by playing them both over separate loudspeakers at the same time that the stereophonic improvement is noticed.

But to achieve this improvement we need two complete recordings — two pickups, preamplifiers, and amplifiers for the whole playback operation, as well as the two separate loudspeaker systems.

Each complete channel carries a high-quality full-range presentation of the program, although they are both very similar, requiring critical examination to detect the all-important *difference.*

If the program is broadcast over the radio, two separate broadcast channels are required to carry the separate stereophonic channels. These may be two FM-channels, or sometimes an FM- and an AM-channel.

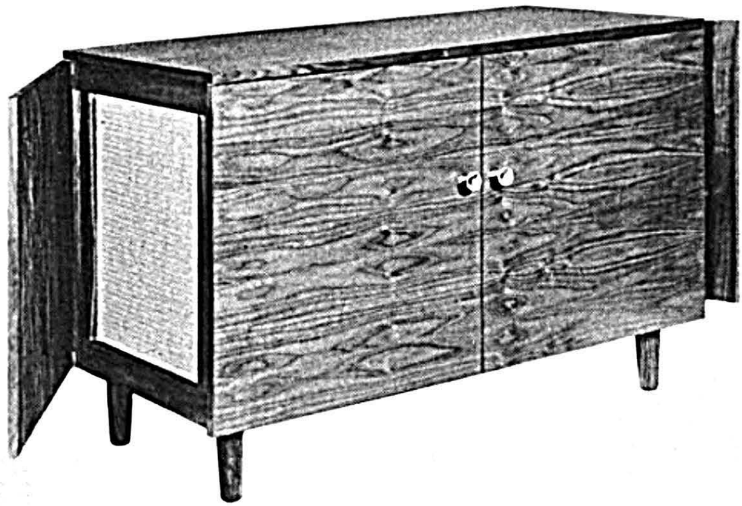
With this system, if the two loudspeakers are played fairly widely apart, according to the shape of the room where the reproduction is presented, there is apt to be a complaint about "lack of center". When the reproduction is intended to represent a source of sound equally distant from two loudspeakers (or in the middle), it is apt to seem somewhat lost. True the impression, if any, is that the sound comes from the middle, but it seems to do so in rather an indefinite way, as if the sound has to go all around the room before you hear it properly.

This effect with two-channel stereophonic has been called a lack of body, because the center, or body, of an orchestra reproduced by this means, seems to be missing. The sound comes over but it has a kind of "hollowness" about it. If the loudspeakers are placed closer together, the lack of center or body is to some extent overcome. One arrangement puts them back to back in the same enclosure, facing outward (Fig. 28). This arrangement is quite suitable for many average livingrooms.

This kind of stereophonic reproduction can be very impressive on demonstration tapes, such as an aircraft flying from right to left or left to right, and similar dramatic presentations. But this is not the kind of program you usually want to live with. If you happen to live near an airport (who doesn't these days?!) you get plenty of aircraft flying over, without presenting it over your reproducer system!

As far as musical presentation is concerned, the two-channel stereophonic gives the best realism on program material with low- to mid-range frequencies — organ music, and orchestral music with the string bass, cello, bassoon, tuba and similar low-to-middle tones predominating. Also the bigger drums can sound quite realistic. This is because the time and intensity differences are apt to be slightly exaggerated as compared with natural listening, a condition arising from the fact that the microphones are further apart than the normal spacing in the human head. This is least noticeable on frequencies where the wavelengths are still at least as large as the spacing between the microphones and loudspeakers.

At higher frequencies, where the spacing between the microphones may amount to several wavelengths, the directional effects may become quite confusing, because the relative intensity and phase differences can multiply up several times the natural amount. If you listen to a recording of an orchestral program with the strings played *pizzicato,* it will give you the impression that there



Courtesy Bozak Mfg. Company.

Fig. 28. Single-enclosure stereophonic loudspeaker cabinet, using loudspeaker units at each end, with doors as reflectors.

are just about twice as many strings being plucked as there really were, because each microphone will pick up the same string program with just sufficient difference in time to give the impression that each string is plucked twice instead of once.

2. Three-Channel Stereophonic

This is virtually the original stereophonic system. It utilizes a center channel as well as the two side channels, which is the only respect in which it differs from the first system. The use of a center channel overcomes the "lack of center" and improves the body discussed under the previous heading.

This system is always used with the microphones spaced several feet apart. Consequently the effect on different kinds of program material is quite similar to that with the two-channel stereophonic.

The best results are obtained on program material in which the dominant frequencies are in the low-to-middle range.

The remarks about each channel carrying almost exactly the same information, with just slight intensity and phase differences, so that any channel could equally well be listened to as a single-channel reproduction, applies to this system as much as to the two channels.

Fig. 29. Best position for loudspeakers with three-channel stereophonic is along one of the shorter walls of a listening room.

Consequently it means the system is just about three times as costly as a single-channel arrangement, to produce its particular approximation to stereophonic sound.

Unless you have a larger-than-average livingroom, speaker placement can be a problem. The ideal arrangement seems to be with the three loudspeakers spaced along one of the *shorter* walls of the listening room (Fig. 29).

3. Stereosonic

Notice that this is a different spelling. It is a system patented by the EMI* in England and utilizes quite a different principle from the two-channel stereophonic. It is the only system that basically uses *directional* microphones and deliberately does not employ a time difference as well as an intensity difference in the material on the two channels.

The pickup arrangement uses two bidirectional microphones, with their directivity pattern at an angle (Fig. 30). The whole 90° between the maximum points of the two microphones can be used as a pickup area for the program material.

An artist playing on the extreme left-hand side will be picked

* Electrical and Musical Industries.

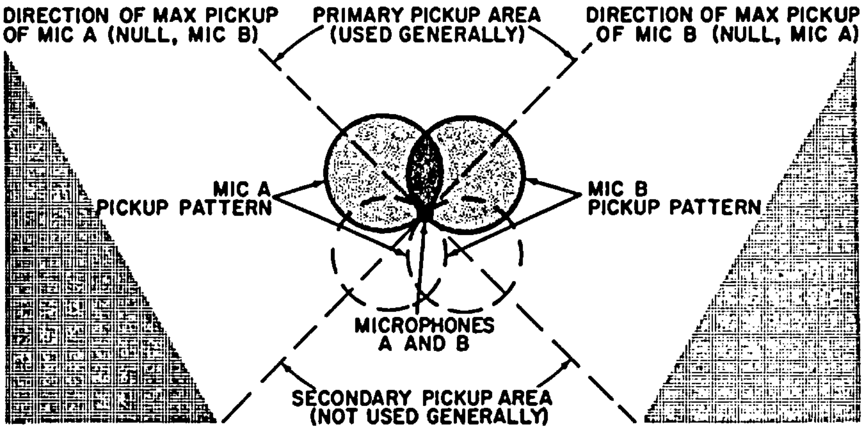


Fig. 30. Microphone pickup patterns for making the stereosonic recordings.

up only by the left microphone, pointing in that direction, because he will be edge-on to the sensitivity of the right-hand microphone. An artist playing at the extreme right-hand side will be picked up only by that microphone, while one in the center will be picked up equally by both microphones.

This means there is a much bigger difference between the program content of both channels, but there is *no phase difference.* The only difference is that the individual instruments are recorded at different relative intensity on the two channels.

This method of pickup avoids one of the chief deficiencies of the two- and three-channel stereophonic — the excessive phase difference between the program content of the individual channels which at the higher frequencies can produce quite an erratic, resulting frequency response of the system, as well as making the directional effect of these upper frequencies quite indefinite. By recording everything on both channels in exactly the same time or phase relationship, there is no possible conflict from this source, any more than there is from just running two loudspeakers off the same single channel.

The paper giving the theoretical foundation for this system explains, with mathematical support, that intensity differences from the loudspeakers produce *effective* phase differences at the ears of a listener anywhere in the room, so as to give a similar equivalent sense of direction practically independent of where the listener may be situated. This is not achieved by either the two-channel or three-channel stereophonic system to the same degree.

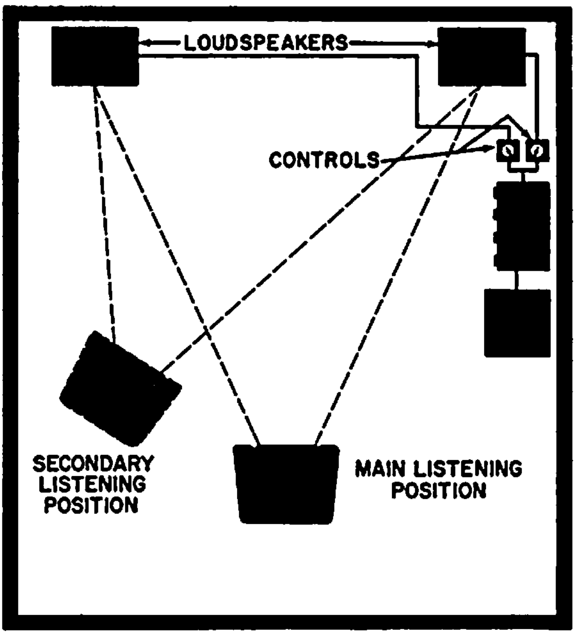
A better way of showing that this principle works (and far more convincing than the mathematics when we remember the discussion we had about the relative importance of continuous tones and transients), is to demonstrate the effect of varying the amount of power fed to two loudspeakers from a single channel. This experiment is illustrated in Fig. 31.

As the control is turned so the loudspeakers receive either equal power, or more to one than the other, the apparent source of the composite audio appears to shift from the center between the two loudspeakers to the left or to the right. This effect is quite definite, although the power fed to both loudspeakers is in the same *time* phase at all times, because it comes from the same channel — a single-channel recording.

What the stereosonic system does is to feed varying intensity combinations of different components of the program (different parts of the orchestra) to each loudspeaker. This means the different parts of the program will come from different apparent positions in the same way as the whole program is shifted by using the control of Fig. 31.

There is another possibility with the stereosonic system. The

Fig. 31. The effect of only loudness difference between two loudspeakers: In the main listening position the best illusion of direction is obtained, but some variation is also noticeable at secondary positions.



paper describing the system showed that the program material picked up by the microphones can be combined so as to produce a form of coded channels. The output from the two microphones is first amplified, and then these channels combined in such a way that one output channel takes the two microphone channels in phase with one another, while the other output channel takes the two microphone channels out of phase (Fig. 32). These two com-

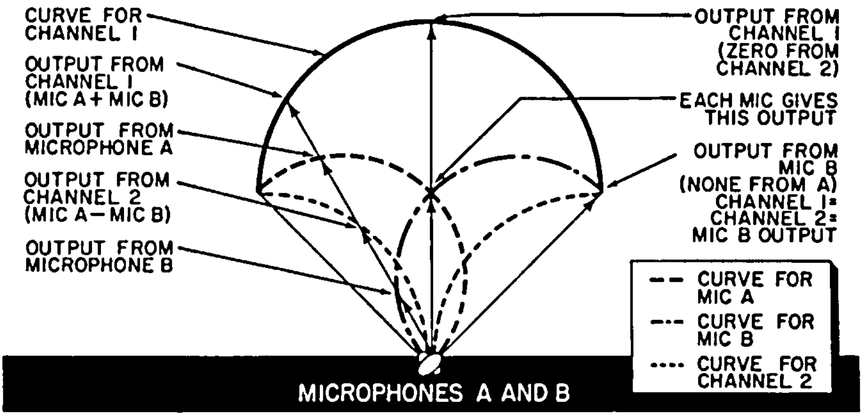


Fig. 32. How the output from the two microphones in the stereosonic system can be combined to give a major channel 1, and a minor channel 2.

binations or output channels then form the program that is recorded, on what we will continue to designate as channel 1 and channel 2.

Any instrument that is on the 45° line, in between the two microphones, will *only* be recorded on channel 1. Nothing will be recorded from this position on channel 2 because both microphones pick it up at equal intensity and there is no *difference.*

An instrument on the most sensitive position of the first microphone (which is the "dead" position of the second) will be recorded on channel 1 and also on channel 2, both in the same phase, we will say. However, an instrument on the most sensitive position of the second microphone will also be recorded on both channels, but this time with channel 2 out of phase with channel 1.

Notice that with this arrangement, channel 1 carries all the program material *in almost uniform intensity,* while channel 2 carries only the program at the two sides, with practically nothing from sounds originating in the middle.

This method has the advantage that channel 1 may be used as a single-channel program and played over a single loudspeaker, maintaining the balance of the original program material. With the original method, neither microphone gives a truly balanced pickup, because an instrument on the maximum pickup position of one microphone is not picked up at all by the other or *vice versa.*

Another advantage of this arrangement is that channel 2 carries only a minimum amount of information and thus does not need such a high-fidelity recording channel as channel 1. This may prove to be a considerable asset of the system. While all the other systems so far described, require – as many channels as are needed – of *uniform fidelity,* this one will give quite successful performance with only one channel of maximum fidelity. The system will give almost indistinguishable performance if the extreme low frequencies are completely missing from the second channel and also if some of the high frequencies get dropped off in playback.

Of course, the channels are recombined in the same way, by taking sum and difference arrangement after the first playback stages, producing recombined loudspeaker channels (Fig. 33).

This method involves a slightly more complicated electronic equipment, but means it should be possible to effect economy in recorded program cost, or programs for transmission over two separate radio channels. The highest fidelity channel, for example the FM, can be used for channel 1, while a relatively low-quality audio channel, an AM transmission, can be used for channel 2, without noticeably deteriorating the quality of the resulting stereophonic presentation.

4. Coded Single-Channel Stereo

At the moment of writing no such system is available for home presentation of stereophonic recording. It is presented here for two reasons: (1) it is known that certain companies are experimenting with a variety of possible systems; and (2), this approach gives at least a theoretical possibility for considerable economy in effective stereophonic presentation.

In its simplest concept, it differs from all other systems of stereo in that only one channel of audio is needed. In the stereosonic system only one high-fidelity channel was needed and the second channel could have quite restricted range, but in this system only one channel of audio is used. To effect different distribution of sound between the two or three loudspeaker channels used, some kind of coded information that is not audible has to be recorded, along with the program material that is audible.

The code information can either be on a separate channel, in which case the channel separation prevents it from being heard, or it can be recorded in some inaudible fashion on the same channel with the audio. The latter, of course, represents a very considerable economy over other systems.

In this case it can use either subaudio frequencies, that is, frequencies that are too low to be audible, both in frequency and intensity, or it can use supersonic frequencies — frequencies too

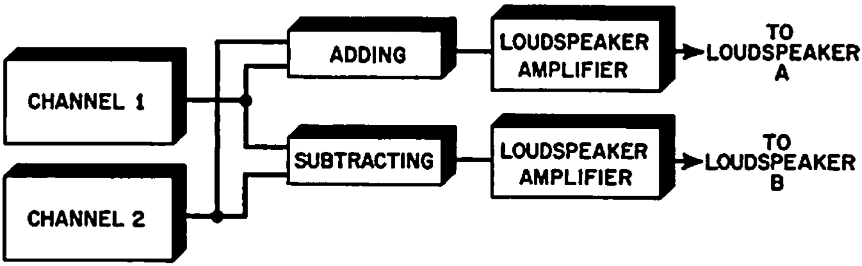


Fig. 33. Recombining on playback with a simple electronic "adding" and "subtracting" arrangement restores the original A and B information picked up by the stereosonic microphones.

high to be heard. Yet a third possibility is that it use a narrow channel of frequency right inside the audible band, but the system takes special precautions to see (a) that the control frequency is not audible, and (b) that program frequencies do not operate the coded control.

The use of supersonic control-code frequencies means that the recording mediums must be capable of maintaining good reproduction up to these frequencies, without any intermodulation from frequencies that are audible. This proves to be a serious restriction.

The last possibility mentioned means that unless the recording is played over the system designed for it, the code frequency will be audible in a single-channel reproduction. This would be a serious drawback, because one of the big advantages of a single-channel, coded stereophonic is that it could be made compatible with existing single-channel systems, so that if a coded stereo record or input were to be played over an ordinary high-fidelity system it should give good reproduction (but not, of course, any stereo effect). Using this third possibility for coding would, at whatever frequency was chosen, produce a peculiar sound that had nothing to do with the music.

This would appear to leave us with the subaudio control frequencies as the most logical solution. They also happen to be most logical in relation to the kind of coding information they need to carry. We will not take up much space on this as the system is not yet obtainable, but a short description of how the method works to achieve a satisfactory stereophonic presentation is included so the prospective merits can be judged.

Such a system can be designed to feed either two-channel or three-channel loudspeakers (or even more, without requiring more *recorded* channels).

In the two-channel variation it differs from the stereosonic in that both loudspeakers carry exactly the same program material at all times but in different *overall* loudness from time to time. In the illustration we used for the stereosonic, the violins could be playing at the left and the drums at the right *at the same time,* and each would be predominantly reproduced by its own loudspeaker. With coded stereo this is not strictly possible.

However, a little thought will show that a program of this type is rather unusual — especially the desire to be able to differentiate between the two *all the time,* through the program. In practice we do not try to hear the different parts of the orchestra separately unless the part we are interested in happens to be playing what is *essentially* a solo — the rest of the orchestra playing merely an accompaniment for the solo part. In these circumstances the composer directs our attention to different instruments in the orchestra *in turn* by his composition of the music. This may be quite a quick succession of transfer of interest from one to another, but just the same it is a transfer. We do not actually try to listen to both at the same time.

In quite a lot of modern music there is a primary melody that is possibly played by the strings, together with a secondary musical theme that appears in the intervals of the primary melody, the secondary theme being played by some other musical instrument, possibly the wood-winds. Coded stereo can adequately take care of this situation. It concentrates the attention on the appropriate loudspeaker each time the strings play the primary, and refers our attention momentarily to the other side every time the secondary theme comes in.

From demonstrations of this system in experimental form it has been found capable for the presentation of a great variety of program material. The impression conveyed is at least as realistic as the ordinary three-channel stereophonic and sometimes more so. For example, when strings play *pizzicato* this system does not produce two or three times as many strings!

The coding works by changing the amplification of the same composite audio fed to each of the loudspeaker channels (Fig. 34).

The system has another advantage: that it will improve the effective dynamic range of the medium used. At present, although high-quality tape-recorders such as those used for professional purposes, can achieve a dynamic range about equal to a disc recording, this is not possible in the low-priced recorder suitable

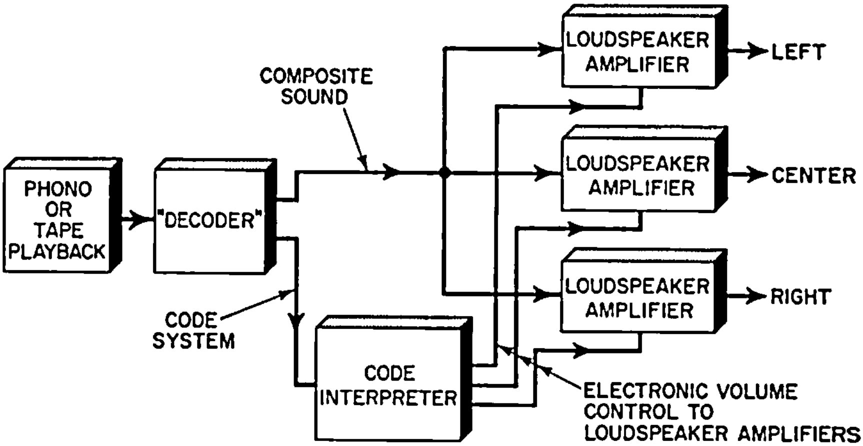


Fig. 34. The basic working arrangement of single-channel coded stereo.

for home high-fidelity use. Consequently, as used for high-fidelity playback, disc recording still gives the best signal-to-noise ratio, or dynamic range. Tape-recorders lag behind in this respect.

The coded stereophonic system helps to overcome this by providing what is effectively a volume expansion. When the program is playing quietly the amplification of all three channels (or two channels, as the case may be) can be turned down so as to reduce background noise as well as the program loudness. When the full power of a *crescendo* is required, the two or three channels are turned up to maximum amplification and thus a greater effective dynamic range is obtainable than with any other stereophonic system.

Stereophonic Program Mediums

So much for the systems available, or potentially available, for use in the home. We now come to consider the different mediums that can be used to present the program material over these systems.

Basically the source mediums, as the home user sees them, are radio, disc and tape. There are other hypothetical possibilities but they probably will not be used for home systems. In discussing each we want to see how they will line up for home use, on the score of:

(a) Simplicity to use: they should not be difficult or complicated to put on.

(b) Economy: recordings should be available at a price suitable for as many as possible to obtain copies.

(c) Dynamic range: they should be able to present the fullest possible dynamic range in recorded material; and

(d) Compatibility with other systems.

This last feature also affects the cost question, because it will mean that recordings do not have to be purchased exclusively either as single-channel *or* stereophonic; a single recording can be issued that will serve both purposes if the method of presentation is compatible.

1. Radio

This is not, strictly speaking, a medium in itself in most instances. Occasionally perhaps, live programs will be broadcast over stereophonic channels. But most often stereophonic presentation of program material will be broadcast from some kind of recording. Consequently the radio is not a medium in itself but a link between the broadcasting station and the listener, and as such is constitutes the input to his system.

It also has an economic factor from a different aspect than the other mediums, because of the limited channel allocations available from the FCC. For two- or three-channel stereophonic presentation the ideal arrangement would be two or three FM-channels as near as possible to the same frequency allocation, so any fading or fluctuation in quality on the different channels keeps more or less uniform, and can be adequately controlled by the AGC and the AFC action of the receiver, more or less in sympathy.

Use of the FM- and one AM-channel for two-channel stereophonic has been fairly successful over short distance transmission where the quality of reception is not subject to appreciable variation. For transmission over even medium distance this method is to be deprecated because of the varying way in which the totally different kinds of transmission will fluctuate in quality, producing some quite spurious stereophonic effects not intended in the program.

For stereosonic presentation the use of an FM-channel for channel 1 of the combined signal, with an AM-channel for channel 2, would be quite satisfactory because of the lesser degree of fidelity required on the second channel.

Similarly in transmitting coded single-channel stereo: this could, by some sacrifice at the low frequency end, be transmitted entirely over a single FM-channel; alternatively, by using an AM-channel of extremely restricted bandwidth and therefore low inherent noise pickup, the combined FM and AM arrangements should produce extremely good transmission results with coded stereo.

In both the stereosonic and coded single-channel stereo transmission over radio, the fact that the important, high-quality audio is all transmitted over one channel, minimizes the possibility of spurious stero effects being introduced by noise or fluctuation in transmission quality.

If there should ever be any extensive interest in stereo broadcasting, to the extent that live program transmission would become a common thing, there is a lot to be said for the stereosonic system of working, with FM- and AM-channels. The stereosonic system has a great advantage of simplicity of control for the audio engineer at the studio. This system requires no "fudging" to get the desired stereo effects.

The two- and three-channel stereophonic require careful monitoring to ensure the desired effects being put over correctly, while the coded stereo requires careful gain-riding by the mixer when recordings are being made. Careful rehearsal and reworking can get an ultimate product that gives the desired results all the way through. But in processing live program there is no opportunity to rework if an error is made, even though there may be opportunity to rehearse the program before the actual transmission time.

The stereosonic system also has an advantage of compatibility, in that the program will sound like single-channel if only the FM is picked up. Coded stereo transmitted the same way would be equally compatible for single-channel presentation. However, stereosonic and coded stereo would not be compatible for one another.

2. Disc Recording

When tape first appeared as a recording medium, many people prophesied that it would eventually out-date discs. So far it hasn't. High quality tape-recording machines, with the best quality tape, can produce a dynamic range far better than the disc records *of a decade or so ago.* Even the early tape-recorders, and tests with carefully developed tape, showed this was a possibility. However, modern development with discs and with improved phono-pickups, has brought us to a stage where a well-recorded master, used with careful pressing technique, can produce a disc with at least as much dynamic range as the best tape.

With both tape and disc, the question of the ultimate in dynamic range seems to be a combination of two factors:

(a) How much care and attention is given to the development of the system and better materials; and

(b) How much you are prepared to sacrifice, quantity- and price-wise, in achieving an increased dynamic range.

For disc recording, it is possible to increase dynamic range by utilizing a greater modulation width – having the groove "wiggle" further. For tape recordings, the maximum magnetic density on the track is set by the saturation density of the tape. But dynamic range can be improved by using either a higher speed or a greater track *width*. Use of a 1-inch track width, with very carefully aligned heads to ensure good reproduction of the extreme high frequencies, would improve the dynamic range by 10 or 20 db on the quarter-inch tape now used as standard. So the argument about relative dynamic range between disc and tape proves to be dependent upon other factors than the simple choice of one medium or the other.

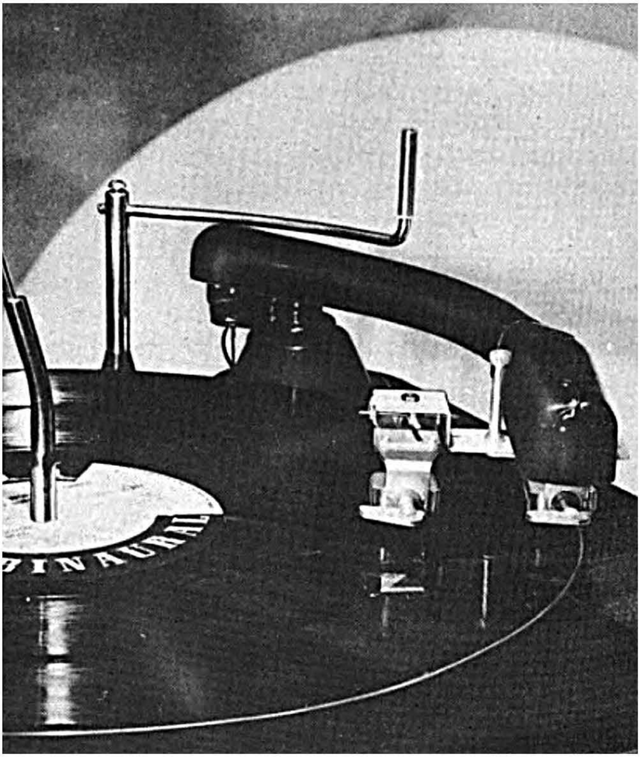
The big advantage claimed for tape is its long-playing time. By using the very thin tape now available, a 7-inch reel will play two hours each way on a half-track recording at 3¾ inches per second. At 7½ inches per second each side plays for one whole hour. This is certainly a considerable improvement in playing time over anything in the corresponding "bulk" on disc.

However, where the disc recording scores – and this seems to be what promises to keep discs in business indefinitely – is in the readiness with which a certain passage can be selected. The modern LP record with from 4 to 6, or even more, recorded bands, arranged with run-on grooves from one band to the next so that if desired the whole record will play continuously, make it very easy to pick out one particular band of a selection at a moment's notice.

Simplicity of use is another advantage of disc. Even the most conveniently arranged tape-recorder requires a little more initiation to use than does a disc-player. But this advantage is about reversed when comparing the Cook binaural disc with stereo tape.

With a tape recording such selection is by no means so convenient. If the tape comes with a footage register for each of the bands, and the tape-recorder has a footage indicator, it is possible to use fast forward or rewind to find a particular place and play just that section. But even this takes a little more time and attention than just placing the stylus at the beginning of the band you want to hear.

Of course, tape shows much improved prospect for several kinds of stereophonic presentation. To apply stereophonic recording, of the conventional type at any rate, on disc, one needs to have



Courtesy Cook Laboratories.

Fig. 35. The clip-on cartridge holder to utilize the two-track type of binaural disc recording.

two or three styli. The Cook binaural record (Fig. 35) has two bands of recording concentrically arranged, with a standard spacing between the pickup styli. The two pickups are carried on a common arm and the record delivers simultaneous output from the two cartridges to the two-channel amplifier.

Some care is necessary in placing the styli in the grooves however, because with the modern microgroove recordings which this uses, the grooves are very close together and some latitude must be allowed so the stylus can adjust itself to compensate for the slight variations in tracking at different positions of the arm. This means it is easily possible for the stylus to be one or two grooves away from the one corresponding to the groove the other stylus is playing.

Naturally the playing time is approximately half, because the number of grooves per inch is on the same order as for regular LP

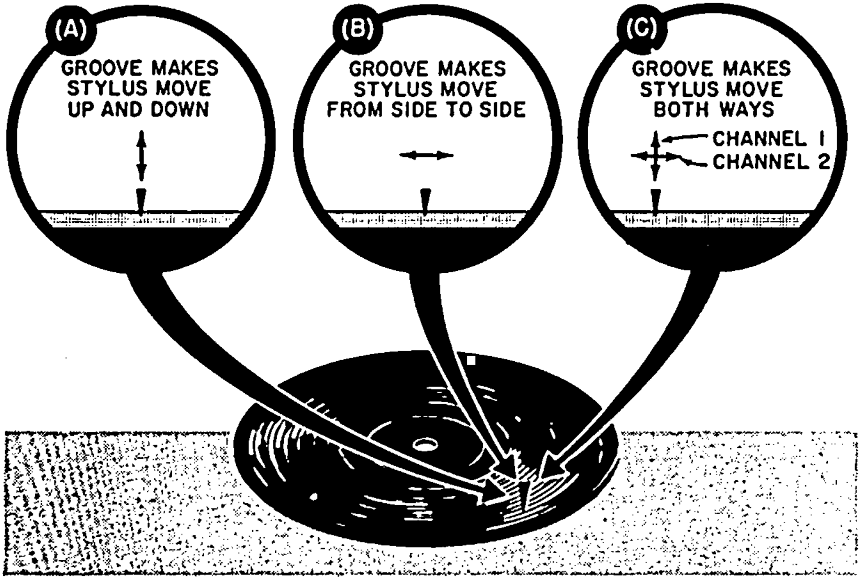


Fig. 36. Different forms of stylus movement used in disc recording: (A) vertical (or hill and dale) used in early phonographs; (B) modern lateral; (C) the experimental single-groove stereophonic uses both.

records, and the space has to be divided between the two bands for the two pickups. Also it is not so practicable to use separate bands as in regular LP's because of the difficulty in getting both styli into the right groove at points other than the beginning of the records. This method of using discs for stereophonic presentation does not seem to offer too optimistic a future. Three-channel would be even more of a problem than two-channel.

The question next arises whether some other kind of disc presentation, using only one groove, might not offer an economic and versatile possibility for stereophonic presentation. Two possibilities might be considered: (1) that all the modulation is contained in a groove with lateral vibration; or (2) that the groove provides movement in two directions, up and down, as well as sideways. The latter method (Fig. 36) would enable twice as much information to be carried in the one groove, and for a long while this possibility has intrigued designers as a means for applying two-channel stereophonics to disc. There are two problems: (1) a very intricate and accurate pickup is needed, and (2) it is difficult to avoid "cross-talk" between the up and down and sideways move-

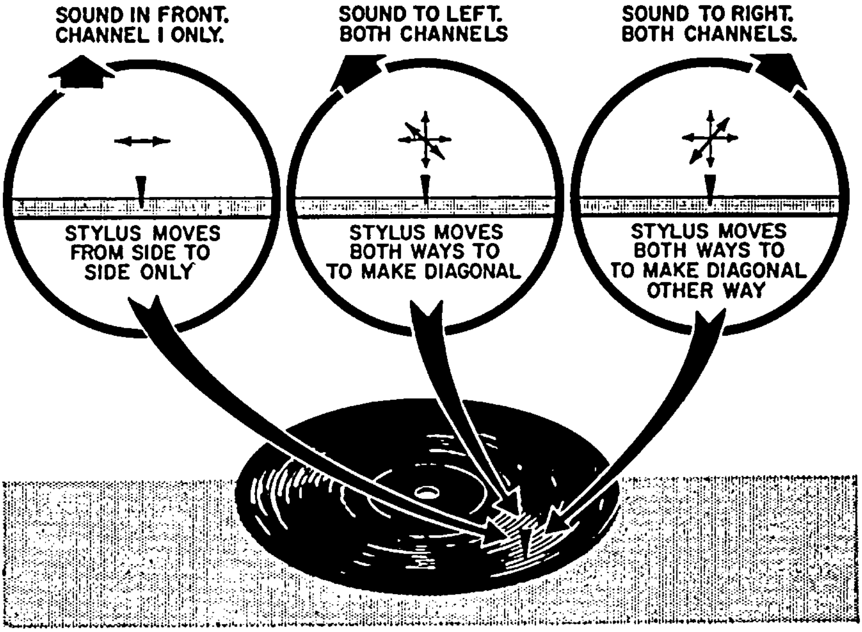


Fig. 37. How use of the stereosonic system, with major and minor channels, fits in with single-groove disc, so the stylus movement reproduces air-particle movement of the stereophonic wave at the microphone.

ment, due to the complicated way the stylus has to move in following the groove.

The stereosonic does offer one improved possibility in this regard. It arises from the fact that the "sum and difference" mixing method puts the main "intelligence" requiring high-fidelity in one channel, while the other channel requires only "difference" information. In this system the lateral vibration of the stylus, as used in all modern phonograph recordings, could be used for channel 1, while channel 2, containing only the "difference" information, could be the vertical one.

This method makes a certain amount of sense on another score: the fact that the stylus movement will now be in exact correspondency with the air particle movement in the vicinity of the pickup microphones. The vertical component of movement one way will combine with the horizontal movement to make the stylus move at 45°. In opposite direction of vertical movement the stylus will move at 45° but in the opposite direction – at 90° to the previous one. This corresponds with the line of movement in front of the original microphone (Fig. 37).

The fact that both channels contain program in exactly the same *time* phase, but in different relative magnitudes, means that any cross-talk that occurs would not produce serious interference effect as with other systems, but will only slightly modify the directional impression conveyed by the system.

It still remains of course, to satisfy condition (1) – to produce a high-quality pickup that will work with this kind of disc recording, and also to figure out a way of pressing such recordings which are somewhat more complicated in the kind of engraving produced, than the ordinary lateral-groove recordings used for modern LP and other kinds of records.

Another possibility is that the vertical movement of the stylus would convey the coding frequencies for the coded stereo. As the up and down undulations would be only at the very lowest frequencies in the audio range, this would considerably reduce the cross-talk problem as compared with any system that requires both directions of vibration to contain the same total frequency range. The stylus movement could be made quite flexible in the vertical direction so that the transmission of the vertical movement does not get into the lateral transducer.

This possibility may make for easier design in the rather complicated pickup needed. On the other hand extra care is needed to make sure the vertical vibration does not get into the horizontal movement, because unlike the stereosonic, in which such a transfer would not produce any form of distortion or unwanted program, this method would produce an unwanted low-frequency buzz.

The alternative way of using a single groove for stereophonic recording requires all the intelligence to be combined in some way, using lateral, or side to side, vibrations only. This means different frequencies must be used for different purposes. If coded single-channel is used, space has to be found somewhere in the frequency spectrum for the coding that will control the distribution of the audible frequencies.

If multichannel is used, only one channel can use the actual frequencies we hear. The other (s) must use supersonic frequencies by conversion (Fig. 38). With this system, accepting a response limited to 12,000 cycles on each channel, a two-channel system would require response to 24,000 cycles. To get three channels into the same bandwidth would require each to be restricted to less than 8000 cycles. As the very high frequencies are very susceptible to damage in a disc, these possibilities are not very practical, because the channel occupying the higher frequencies would be apt to disappear altogether. Another disadvantage is that they are not compatible, because such a disc could not be played on an ordinary system.

Suggestions have been put forward for other methods of making

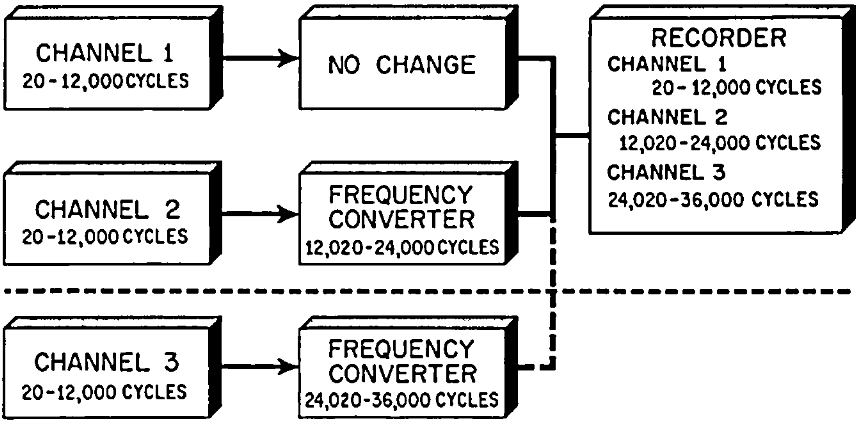


Fig. 38. Multichannel single-track (or groove) can be achieved by frequency conversion, but this method has several disadvantages (see text).

three-channel stereophonic on disc, but they are all too complicated to be regarded as practical, so we shall not go into them right here.

3. *Tape*

The great advantage of tape as a medium for stereophonic recording is that it is relatively simple to put as many tracks as we want on tape. If we could not get three tracks comfortably into a ¼-inch tape, the width could easily be extended. In practice some good three-track tape has been produced within a ¼-inch width.

The principal problem is in the design of heads of sufficient durability and sensitivity to give good dynamic range. Three tracks on the tape require to have two spaces between them, as against only the single space between two tracks. Consequently quite a large proportion of the tape is unusable because of allowing sufficient spacing to prevent cross-talk. The use of narrower track means the output from the playback head is that much lower for full magnetization of the tape. This means the dynamic range is reduced.

Apart from this, use of stacked heads (i.e., with all the three tracks in line) means that the air gaps have to be in line and consequently the head windings need to be very close together. This means only small windings can be used, and consequently it is more difficult to produce an efficient head. Use of staggered heads improves the possible efficiency of the pickup heads. This is an alternative system, where the tracks are spaced apart lengthwise

Fig. 39. The difference in position of recording on the two tracks for stacked (A) or staggered (B) heads.

so the heads running along the different tracks can be separated (Fig. 39).

But the use of staggered heads makes it still more important to avoid cross-talk between channels. Cross-talk between channels of a stacked head would merely mean that some of the program intended for one loudspeaker would break through into the channel for another loudspeaker. Using staggered heads however, means that break-through of this nature results in program getting from one loudspeaker channel to another, *but at a different time,* producing some very spurious reverberation-like effects. Consequently adequate track spacing, and the use of only a very narrow track must be very carefully followed to avoid this kind of thing happening.

Tape is also a very suitable medium for either stereosonic or the coded single-channel stereo. The code can be carried either on the same channel by the use of filters (Fig. 40), or it can be run on the separate track. A possibility here, that may be worth considering to improve overall dynamic range, is to use a somewhat wider track for the audio and a narrower one for the coding. This would improve the possible performance of tape-recorders, and make them easier to produce at a lower cost, because adjustment to get a good dynamic range without excessive background noise and hum would be much easier to achieve.

A final advantage for tape as a medium, at least in a temporary sense, is its excellent possibilities for doing experimental work. For any of the disc systems the whole thing is entirely in the region that must be explored experimentally by professional people — companies with laboratory facilities. But the use of tape as a medium brings the experimental work into the scope of the average home user. Extra tape heads can easily be bought as separate items

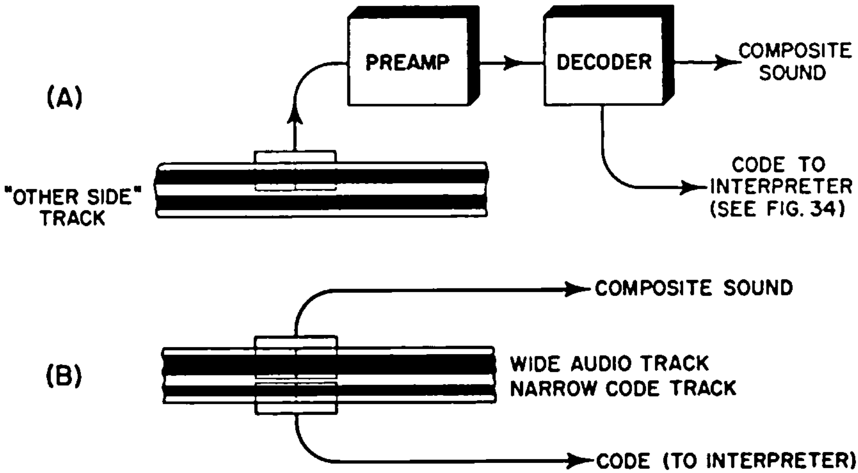


Fig. 40. Two possibilities for coded stereophonic on tape: (A) using one track only, enabling the tape to be used "two-sided" as for single channel; (B) using two tracks of different width to achieve much superior quality.

and mounted up on the deck of an existing tape-recorder. Extra electronic equipment can be built to produce necessary coding frequencies or extra amplifying channels, so the home user can easily build up his own system and try out different variations and combinations that may appeal to him.

4. Other Possibilities

When a satisfactory basic system has been developed by necessary experimentation on tape, it is then not impossible that the same basic system would be transferred to disc for the greater convenience which this medium gives — that is, if it is possible either to condense the total program material into a single channel so a lateral groove could be used, or into a form suitable for lateral and vertical at the same time.

Ever since the advent of recorded program material, other methods of making the recording have been explored. While to date the only popular forms are disc and tape, there are others in the experimental stage that may still ultimately come to the fore again.

Wire of course, was another form of magnetic recording that was popular before oxide-coated film in the form of tape took its place. Tape is so much more manageable that the use of wire has become almost extinct. Wire is very difficult to handle, especially if it should break, when it inevitably becomes tangled up in the most impossible manner.

Another possibility that has been explored is different kinds of optical film. Various sound tracks as used in the motion picture industry have been tried out from time to time for home phonograph purposes. Unfortunately, none to date have proved to be competitive, either in quality or cost.

An alternative system that uses a very similar principle is the Miller system. This does not use optical recording but employs optical playback. The recording is achieved by means of a wide-angle cutter on a film base consisting of transparent plastic coated with a black layer. The wide-wedge cutter plows into the plastic and removes a variable width of the black layer, producing a track very similar to the variable-area track made optically. This system has the advantage that the photographic processing is unnecessary, and the cost of the cutter-head is probably lower than the complete optical system necessary to produce optical recording.

Playback, of course, is almost identical to any optical playback arrangement, so for the home user buying a complete recorded program to be played at his leisure, and not particularly interested in making his own recordings, it is immaterial whether a recording is made optically or by the Miller system. The playback requirements are the same.

Chapter 4

RECORDING PROCEDURE

As soon as you start to talk to recording studio engineers they all express one thing in common. Their other expressions on the subject of stereophonic recording may differ quite widely. But what they all say is that sometimes a stereophonic recording will come out beautifully while at others the thing seems to be a complete flop. This, of course, is quite apart from the kind of faults that can develop in any recording session.

If you pursue matters, expressions will differ. By "coming out beautifully", some will say the production is "realistic", while others freely admit they are not looking for realism, but for a satisfying impression, or in some cases an impressive satisfaction!

Different Schools of Thought

Some studios have used pressure, or unidirectional microphones exclusively, partly on the theory that the human hearing is a pressure-sensitive device rather than a velocity-sensitive one, and that the directional faculty is achieved only by comparison of what two separate ears pick up. Consequently, based on this theory, these recording studios have worked entirely with pressure-type microphones spaced apart at various distances.

This kind of technique leads to a preferred position above the performers and slightly to the fore. This is necessary because keeping on a level with the performers, it is impossible to get close enough to them to get (a) sufficient discrimination against an exaggerated reverberation effect, and (b) sufficient differential between the two or three channels. The only way to get "close enough in" without being too close to some, is to use an overhead position. Our comment would be that using a "natural" pressure microphone evidently necessitates an unnatural position!

Some studios on the other hand, have expressed themselves as finding directional microphones – either of the cardioid or the bidirectional pattern – much preferable. In most of the larger studios with normal reverberation characteristics – that is, characteristics usually associated with rooms or auditoria of this size – a cardioid gives the best results, because the back pickup of a bidirectional ribbon still makes for a somewhat higher reverberation than seems natural.

The best way to use a ribbon successfully in these circumstances is to completely rearrange the performers so that back and front are used as live pickup directions, and everyone is just a little nearer to the microphone than they would be if all were arranged on the same side (Fig. 41). This introduces another factor encountered by the recording studio engineer.

Musicians are Important, Too!

This is the extreme desirability of not interfering with the musicians' positions or playing technique more than is absolutely necessary. Musicians are people with a long background of experience playing in orchestras, using a more or less normal layout that has conditioned each one's hearing to a certain musical perspective in his work. Each one can only produce his best performance when he occupies as nearly as possible, a position that gives him this "conditioned perspective".

The conductor is accustomed to placing the musicians in a position that suits him. Other musicians are used to a comparatively limited variety of positions, dictated by various conductors with whom they may have played. If the conductor is to get the performance from his musicians that he expects, he must use his well-proven techniques. To give him a position with musicians placed fore and aft in different directions to suit a peculiar microphone pickup pattern, would completely throw him from his inculcated ability to achieve tonal balance between the different sections of the orchestra.

The sounds he normally hears from specific directions will now be coming from completely new directions, at least as *he* hears them, although the microphones may get the correct overall impression for the two channels recorded. An arrangement that gets correct balance in pickup, but disturbs the musicians so they cannot give their best performance, loses more than it gains.

The same argument opposes the use of studios with deader-than-usual characteristics. Some of the recording companies have developed studios of normal size but much shorter reverberation time. This makes it much easier to use whatever type of microphone and microphone placement may be desirable for achieving the balance or stereophonic effect required, without having to consider what the reverberation effect will be. Then whatever reverberation is

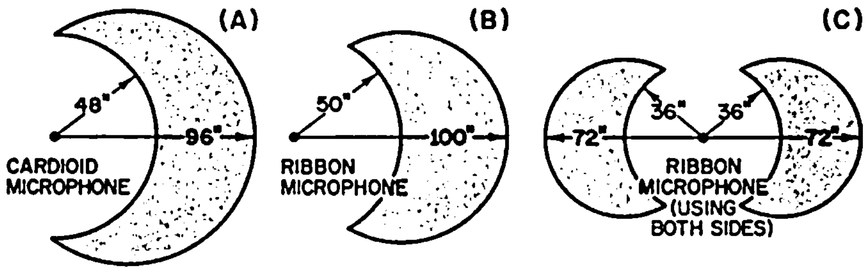


Fig. 41. Comparison of distances from microphone necessary to cover the same total area with (A) cardioid microphone, (B) ribbon, using only one side, and (C) using both sides of the ribbon.

needed can be added by means of a special reverberation room or echo chamber.

This makes the recording engineer's job very simple, except for one thing: the musical director has serious difficulty in handling the musicians and their conductor, because these artists find the acoustic environment extremely unnatural to work in, and they cannot give their best performance under conditions so strange to their aural conditioning. Some engineers have dubbed musicians crazy, hide-bound, fussy and similar adjectives, because they adopt this "difficult" attitude. Correct environment is as *necessary* for a musician to give his best performance as good microphone placement is to make a good recording of it.

Further, the rather-dead studio with make-believe reverberation is an *artificial* method of achieving a close approximation to the intended illusion, without simulating the apparent conditions at all. This can hardly be called "realism"!

Another approach is to use more or less normal studio characteristics but bring the microphones much closer in to the artists, and if necessary use a greater number of microphones, almost one per artist, perhaps. This method uses a general pickup microphone — or three or four — and then adds individual microphones to pick up solo instruments — or groups that are important to the overall effect. The latter are operated at a level to dominate the general pickup.

This method of achieving tonal balance, and a balance between the direct sound from the performers and the reverberation, has been used extensively for single-channel recording. But some of the stereophonic proponents regard this as being too "artificial" to apply to the new medium. We should eliminate all the "trickery" and get back to "genuine" recording, now that we have a stereophonic medium to give "exactly the real thing".

Those who adopt the latter attitude appear to be the very ones who most often find that their results don't come up to expectations. It would seem that when they do come up to expectation, it is something of a fluke, rather than because they have conditions more accurately correct.

Some companies have done a considerable amount of work with stereophonic recording and have gone to almost endless pains trying different types of microphones and microphone placement. It must be conceded that the presentation should be as realistic as possible although we cannot expect identity with the original, and that we *should* be able to establish an ideal microphone arrangement for each session with some theoretical "justification". But suitable theory has yet to be completely defined.

. . . And The Composer

For this reason a clear appreciation of the principles stated in the earlier part of this book should help to improve the approach to stereophonic recording. First let it be realized that there is no such thing as a single correct microphone placement pattern, for stereophonic recording in a given studio. There is not even a correct microphone placement for a given orchestra playing in that studio. Microphone technique has been found to be dependent, among other things, upon the precise piece of music the orchestra is playing — particularly upon who the composer is.

This statement may at first sound heretical, but a brief consideration will show there is foundation for it. Each composer has taken the variety of musical instruments available to him and orchestrated his composition in such a way as to produce the sound effects he intends. Different composers have characteristic ways of using the musical "tools" of the orchestra.

Some use them in successive order, playing one group and then another, so that groups have a separate entity. Others make the different groups play simultaneously, but still with individual parts or characters. Yet other composers hear the sound as a complete merging whole, and the music does not benefit by having the different instruments "separated"; the composition was never designed that way. Differences in the composer's use of orchestral instruments considerably affect *what* stereophonic sound can "do" for his music, and consequently what microphone techniques will best serve the purpose, or *how* stereophonic sound can enhance the particular presentation.

Perhaps, in the future, composers will write music especially for stereophonic reproduction. But music written in the past has been intended for live performance in specific surroundings. Musicians and engineers should work together to create the best possible illusion to follow the composer's intentions.

Monitoring and Playback

Listening to an orchestra in an auditorium our hearing faculty can hear the musical instruments somewhat in stereophonic perspective, but mainly the quality is due to the separation between the direct sound coming from the musical instruments and the reverberation coming from all around the hall. Whatever microphone pickup arrangement we may use, it is not possessed of anything like a human brain to differentiate between the individual sounds that each microphone picks up, and convey the overall result of the sound to a recognition faculty that separates direct sound from reverberation before transcribing it for magnetic or electrical recording. We cannot have such a simple transfer of "intelligence" as occurs in the human brain, listening to a large performance. This being the case, we must endeavor to create the most successful illusion for the particular musical performance in hand.

In the recording profession, some idealists are endeavoring to adhere to a particular technique without modification, placing microphones in specific positions, adapted as little as possible from the theoretical ideals with which they started out. With some performances this can be quite successful, but the success is more by accident than design.

Those without any inhibitions due to preconceived notions as to how it *should* be done, who thus leave themselves free to experiment, have discovered that different microphone arrangements will give best results, even in the same auditorium and with the same orchestra similarly placed, when different musical compositions are being handled. These are not the only differences . . .

The next difference we consider is one which can be a real source of confusion, the fact that the best microphone placement depends not only on the sound being picked up, but also on the manner in which the channels recorded will be played back or reproduced. During recording the engineer and musical director use a 2-speaker or 3-speaker system in a monitoring room to judge how the recording will sound when it is played back. The monitoring room is built to be as near as possible representative of an average livingroom in which the recording is intended to be played.

But the loudspeaker placements can differ. It may be one of the double-ended loudspeakers, or two separate loudspeakers at different spacings along a wall. In the same way that loudspeaker positioning will modify the effect of a specific recording on playback, the positioning chosen will also modify the choice of microphone position to get the best results with this particular loudspeaker placement.

Consequently a recording produced to get ideal results for example, from a loudspeaker consisting of a single enclosure with the units mounted in each end, will not give such good results when played over a system where the loudspeakers are mounted in the corners of the room. Correspondingly, a recording made in which the monitoring arrangement used loudspeakers either in the corners of the room, or spaced a little distance apart, will probably not sound so well over a loudspeaker in which both units are mounted in the same enclosure.

As each studio has standardized on the loudspeaker placement for monitoring, a result of this will be that the user will probably prefer recordings by a studio that uses a monitoring loudspeaker placement most like the loudspeaker system in his own listening room. This will undoubtedly lead to arguments and conflicting preferences as to which studio turns out the best stereo recordings.

Recording Technique

Some studios allow themselves a little more leeway in making up the final masters for ultimate release, by recording the original program on more than the final two or three channels. Some use four, five, six or more separate channels to record the original program, and then work on these in the laboratory, combining them in various proportions until the most satisfactory illusion is achieved. While this procedure does save time for the set-up and occupation of the musicians, all of which is quite expensive, and allows the

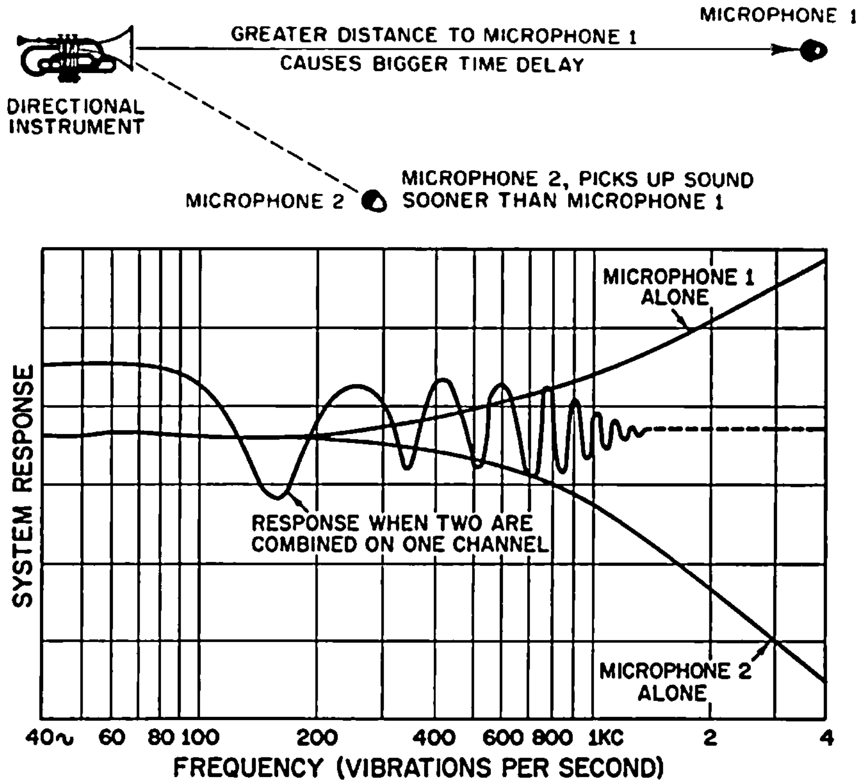


Fig. 42. The danger in mixing more than one microphone that picks up the same instrument is the kind of erratic frequency response shown here.

delicate work to be done more cheaply by a few engineers working at their leisure, it has serious limitations. There are definite limitations to the way in which several channels can be combined to make one or more common master channels that are satisfactory.

Some studio engineers express a preference for using as few microphones as possible, and the reason is not difficult to see. When more than one microphone picks up sound from one or more of the instruments, the combined effect on the ultimate recording will have a phase difference between sound that is brought together in the same channel. If all the separate channels are reproduced over separate transducers, a spatial effect of some kind can be achieved. At least cancellation or partial cancellation at some frequencies, and augmentation at other frequencies due to phase difference, will not occur.

But combining separate channels into one, with phase or time differences from individual components in the musical program, will introduce an artificial "liveness" to the reproduction that is not natural. It will produce peaks and valleys in the effective frequency response of individual musical instruments, in spite of the fact that the individual microphones may have extremely smooth frequency response (Fig. 42). This is why some engineers prefer to use not more than two to four microphones (in the latter case two on each channel), to make the original recording.

However, the opposite technique, using microphones for each soloist or each group of instruments, avoids the problem for an opposite reason. Each microphone only picks up a comparatively small "area" of sound at the final intensity. Any stray-over from other areas will be at a much lower level, so it does not interfere with the sound picked up by the local microphones concerned.

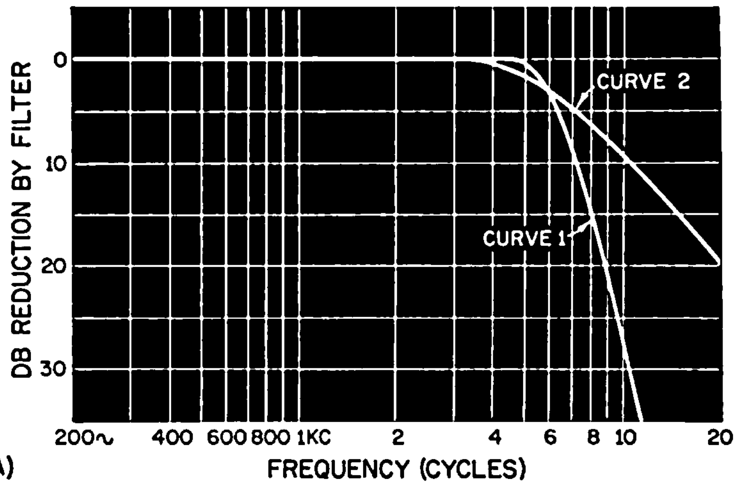
The trouble with using a large number of comparatively local or close-in microphones however, is that the reverberation is reduced to an unnaturally low level. This can be added, it is true, by the use of an echo chamber, but then the result is even more artificial.

For this reason the most commonly accepted method uses microphones fairly close in, but not too close, so as to use a studio of more or less normal reverberation characteristics. The apparent reverberation on the ultimate recording is governed by the spacing between the microphone and the sound source — the musical instruments. A position closer in will reduce the effective reverberation by increasing the relative sound from the sound source. A position further away will increase the apparent reverberation.

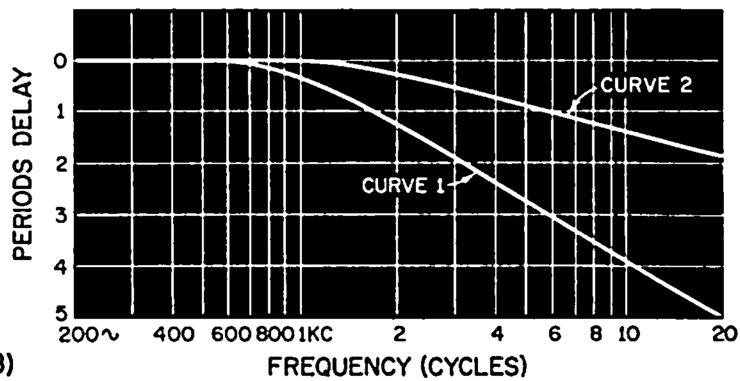
For this reason several studios are finding it preferable to utilize microphones with some directional characteristics, either of the condenser type or one of the cardioid variety. This enables fewer microphones to be used further away from the performers, while still getting an acceptable relationship between the original sound and the apparent reverberation in the final recording. The overall result achieved this way tends to be more realistic, although it may offend the academic susceptibilities of some people who feel that omnidirectional pressure-type microphones only, should be used.

No professional studio recordings are made using the binaural microphone technique with two separate microphones placed in a dummy human head (Fig. 4). This is intended for home binaural recording and similar applications, rather than for serious professional use.

Stereophonic recording is still a new field. Talking to recording engineers from the various studios, one gets the impression that as



(A)



(B)

Fig. 43. The use of a relatively sharp filter to cut off frequencies above a certain point (6000 cycles shown here) produces greater interference with frequencies below this point than a more gradual filter: (A) compares the measured response; (B) shows corresponding delay times. Curve 1 indicates a sharp filter; 2 indicates a more gradual one.

yet, a great many poor tries have been "buried" for each one that has proved satisfactory enough to release.

In discussing stereophonic theory with different recording engineers, one meets some conflicting opinions as to the nature of stereophonic sound, and also the pertinent factors for satisfactory binaural perception. Many independent investigations have been conducted for example, into the effect of frequency range, or the effect of different sections of frequency range, on binaural perception of the location of individual instruments, both "laterally" and in "depth", from the reproduction of two loudspeakers.

Conclusions from these varying sets of experiments seem to disagree, not because people with different kinds of ears were listening, but because the explanation of the results has been somewhat faulty. Unfortunately, many are still hypothesizing on the basis of steady tone composition – discussing intensity and *phase* differences. This can lead to some faulty conclusions.

For example, to determine whether frequencies above, say 6000 cycles, are necessary, a low-pass filter which allows all frequencies up to 6000 cycles to be amplified, but removes frequencies above this point virtually completely, is used to compare the effect of recordings "with or without" the top range, from 6000 cycles upwards.

On such a test it seems that these high frequencies are vital to proper stereophonic perspective. However, a factor that seems to have been overlooked in this particular experiment, is the affect of such low-pass filters upon the *transient* response of the system. It has been proved, both theoretically and experimentally, that any filter with a sharp cut-off that suddenly ceases to amplify above a certain frequency, also produces *time* differences between the frequencies well below the cut-off point.

This means that frequencies in the region of from 1000 cycles up will have the time relationship of transients modified by the use of such filters, although the response magnitude-wise, may be quite level (all the frequencies are reproduced in proportionate amplitude). But correct binaural perception is dependent upon correct *timing* of all the frequencies heard, as well as correct magnitude. In fact, other things being equal, it is more dependent upon correct timing. Notice here that we are concerned, not with the correct phase relationship *between* the two channels, but the correct *timing* of the individual frequency components of the waves from each channel. As the sharp cut-off filter upsets the timing of frequencies from 1000 cycles up, it is understandable that addition of these filters would considerably mar the stereophonic illusion.

Had a filter been used which did not cut off so sharply, so as to destroy the timing of frequencies below the cut-off point, quite different conclusions would have been reached (Fig. 43). Experiments conducted in this way show that the high frequencies are less necessary to stereophonic illusion than the mid-range, up to somewhere between 6000 and 10,000 cycles, as has been discussed elsewhere in this book.

This once again underscores the importance of thinking about stereophonic effects in terms of sound *impact* rather than frequency response.

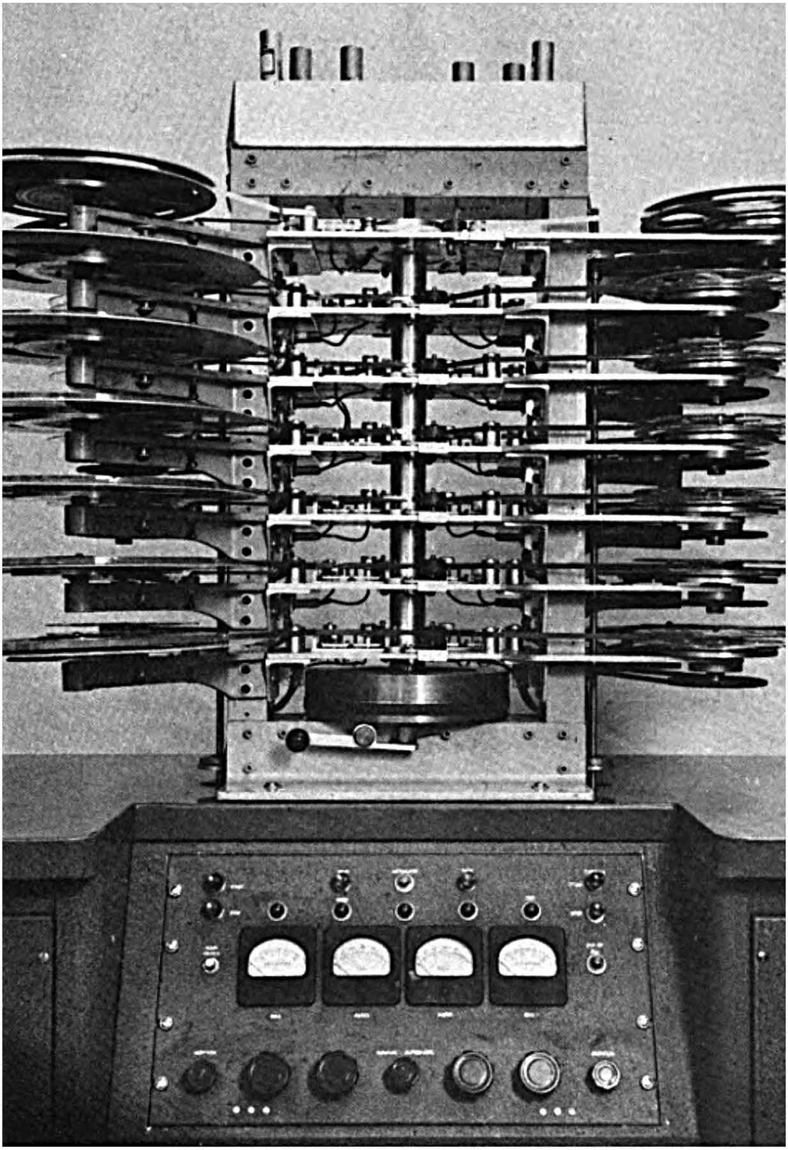
Another question connected with record-making is the kind of medium to be used for final release of the recorded program, whether to use tape or disc. So far, of course, the only available disc method uses two separate tracks spaced apart radially. However, the idea of using both vertical and lateral movement of the same stylus has intrigued a number of people for some time and experiments are going on with a view to developing this system.

The fact that magnetic tape does not use any moving parts, and thus avoids the mechanical complications bound to be involved in any two-way pickup, makes most people in the industry confident that magnetic tape is really the ultimate solution. The big appeal for disc, however, is its extreme versatility.

On the other side of the score, versatility is being improved for magnetic tape by the development of tape cartridges. These will save the necessity of threading the tape through the machine, which at the moment is a job for only one member of the average household — the owner of the tape-recorder. Almost anyone however, can put on a disc.

Years of work have already gone into the production of satisfactory disc techniques, whereby discs can be pressed quite rapidly. On the other hand, pre-recorded tape production is a relatively recent development. Even the quality of magnetic oxide on the tape has only recently developed to its present standard, and it is highly probable that it will go on improving for some time to come. This may result in tape ultimately being a medium of much better quality than discs could ever be, especially in view of the fact that disc depends upon a mechanical stylus vibration for the reproduction.

The problem in tape duplication is that of speed. There is no counterpart to the pressing method, whereby discs are produced, stamping them out in rapid succession. However, multiple duplicators have been designed capable of running the tape through a number of heads simultaneously at many times the actual playing speed used for reproduction. This enables the transfer to be effected much more quickly, and a great many tapes to be printed at the same time (Fig. 44). Further development in this direction may yet yield a tape duplicating process that competes favorably as to ultimate cost to the consumer, with the pressing method for discs.



Courtesy R. E. Marshall and Dubbings Sales.

Fig. 44. A modern multiple tape-copying machine. With high precision built into it, and a much higher bias frequency, this machine can copy at several times the normal playing speed, without losing original quality.

Chapter 5

LOUDSPEAKERS FOR STEREOPHONICS

What has gone before should give a good picture of just what goes to make a first-class stereophonic presentation, so that now we are in a position to deduce the features to look for in loudspeakers, when selecting them specifically for stereophonic presentation.

General Type

In discussing the types of loudspeaker used to present different types of program material, we commented that the larger, multiple-unit loudspeaker, particularly in a large room, is best suited for presenting orchestral program, because it gives the impression of a large distributed source, even though, as we have discussed, this distribution is not correct — it is by frequency rather than by the different instruments in the orchestra. People who are interested primarily in the reproduction or good orchestral music have acquired such a system for single-channel presentation, and then added to it another unit for two-channel stereo (and sometimes even a third unit for experimental three-channel stereo). They have then expressed surprise because presentation of the stereophonic program over their two- or three-way loudspeaker system does not sound any more impressive than, if *as* impressive as, single-channel, also distributed over the same two or three loudspeakers.

On the basis of this kind of experiment more than one person has come to the conclusion that stereophonics is just so much balderdash and poppycock. Better results they find, can be obtained from single-channel presented over two or three loudspeakers. If this is the kind of loudspeaker you use, the conclusion is perfectly true.

One purpose of stereophonic presentation is to correct the deficiency in using this kind of loudspeaker as an alternative means of getting "depth".

The defect of this kind of system is that the sound is not distributed position-wise according to instruments, but according to frequency. Having two or three of these loudspeakers producing at once will not correct this defect, although it may add to the sense of depth; then, using stereophonic program material presented over two or three loudspeakers of this type will simply add to the confusion of sound presented.

This points up a basic fact about loudspeakers for stereophonic presentation. Whatever the kind of program material, they must be well-integrated, so that each loudspeaker gives a good impression of its source point, and does not itself sound like a spread-out source. The same remarks that apply to selection of loudspeakers for single-channel high-fidelity use, apply in some measure here, i.e., that a loudspeaker that sounds well-integrated in a large room, may show lack of integration in a much smaller room. This is because the listener will inevitably be much closer to the loudspeaker.

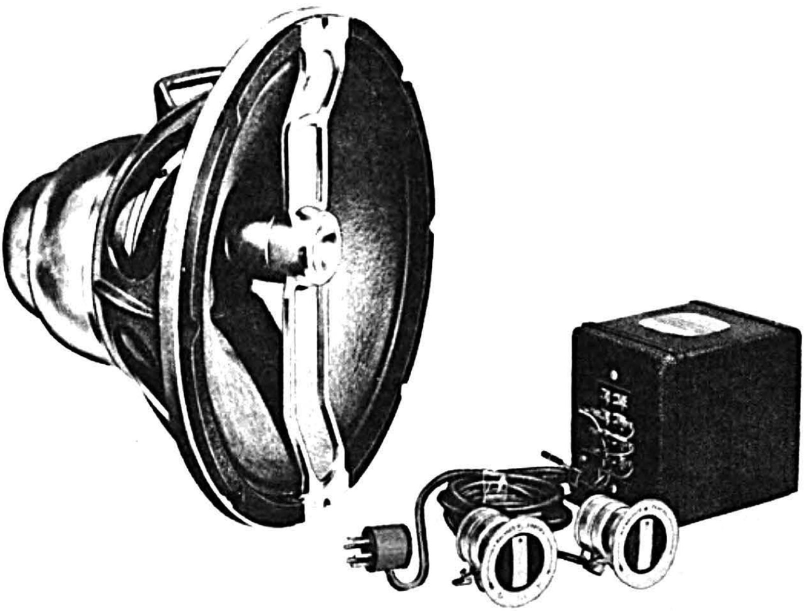
Possibly the two or three loudspeakers referred to at the beginning of this chapter would give very good stereophonic impression if distributed on the stage of a large auditorium instead of in a livingroom. Under these conditions the listening room would be so much larger than each loudspeaker unit, that even these well-distributed sources would sound integrated.

For rooms in the livingroom category, however, the best type of loudspeaker to use is one of the concentric or unitary type, where all the loudspeaker units radiate on the same axis (Fig. 45). Most of these units come rather expensive. If you don't want to pay as much, you may well get very good results with a simple extended-range loudspeaker, coming in a single unit (Fig. 46), and not requiring a crossover.

Frequency Range?

"But surely this will show noticeable loss of fidelity in comparison with a better unit?", some readers will ask. The fact that on stereophonic presentation the loss is as little noticeable as it is, has been a cause of surprise to many. But the understanding of stereophonic sound and binaural listening given in the earlier part of this book, explains it quite simply.

We are primarily concerned in creating a satisfactory illusion of realism. The more nearly the reproduced sound duplicates



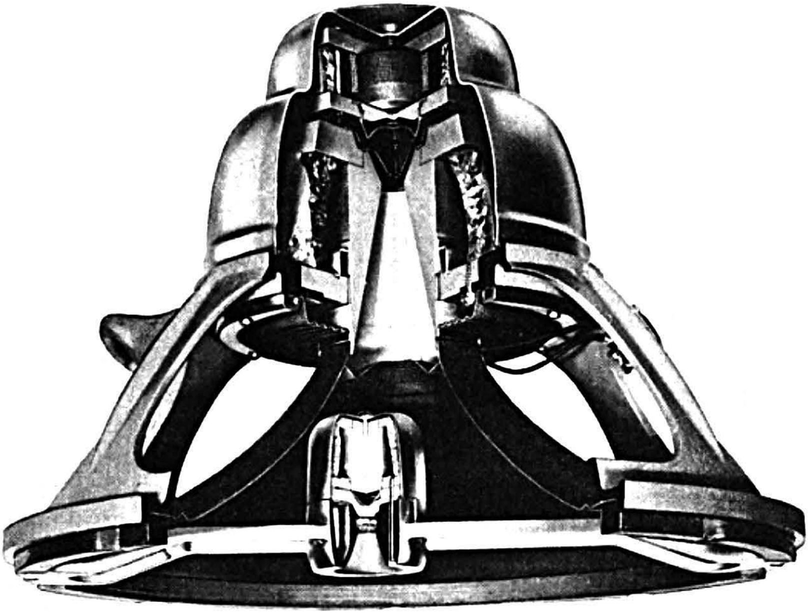
Courtesy Jensen Mfg. Company.

Fig. 45A. The Triaxial: A good example of a three-way, well-integrated loudspeaker unit.

the impression we would have gained listening to the original program, the better the sense of realism we get. The fact that a good stereophonic program presented over two or three loudspeakers gives a sound distribution pattern corresponding better to the original, means that our impression of realism is considerably improved.

When we only had one loudspeaker operating on a single-channel of recorded sound, we relied on the fidelity, provided by the higher frequencies particularly, to convey our sense of realism. It was the accurate rendering of the different overtones providing the timbre for the different instruments, that gave us our sense of realism. Good low-frequency response is also necessary to provide body.

But now that we have two- or three-loudspeaker units operating on separate program material, body is adequately provided, because even in stereophonic presentation the low frequencies are all relatively in phase, while the stereophonic presentation takes over in a better way the function, previously served by the presence of



Courtesy Jensen Mfg. Company.

Fig. 45B. A cross section of the Trioxial three-way loudspeaker unit.

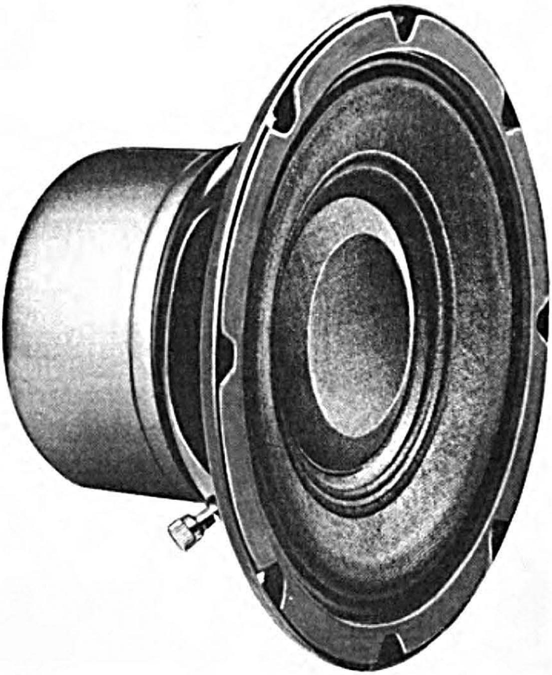
the extreme high frequencies, of giving definition and character to the individual instruments and parts of the program source.

To confirm this principle, it has been effectively demonstrated that loudspeakers with a restricted frequency range, when used on stereophonic presentation, sound to have a better frequency response than they do on single-channel presentation. Just thinking in terms of engineering it was difficult to see why this should be. It would seem that nothing could make frequencies appear to be there when in reality they were not.

The previous discussion shows that the function of the higher frequencies is not to be heard for themselves alone – they merely add character to other sounds *that already are heard,* whether the high frequencies are present or not. These extreme high frequency components make the sound we already heard seem more accurate. As stereophonic presentation serves the same function as far as our listening is concerned, it gives us the impression that the high frequencies, *upon which we previously depended for this intelligence,* are present, although in fact they may be missing.

This does not alter the possibility that the use of a loudspeaker with a wider frequency range extending out to the limit of audibility, may still improve the overall reproduction. The point to appreciate is that the apparent improvement is much more marginal than it was with single-channel high-fidelity presentation.

There is another reason why this marginal property of the

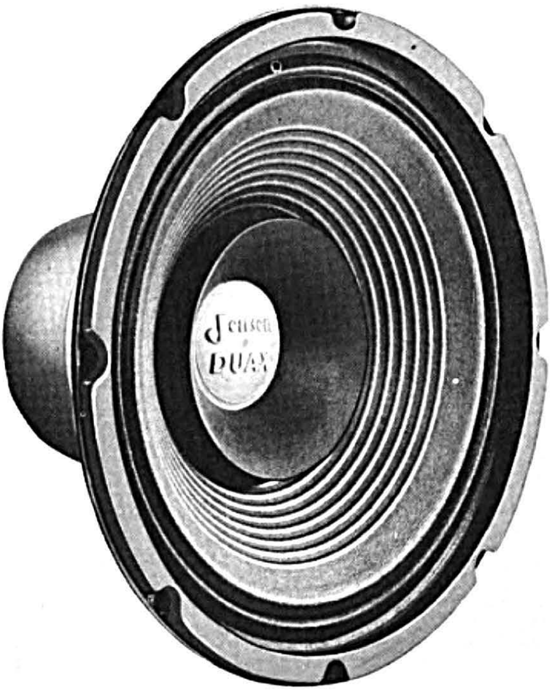


Courtesy Jensen Mfg. Company.

Fig. 46A. The Unax: An example of extended range loudspeaker design.

extreme higher frequencies can become even less of a benefit, however. While the original sound sources are integrated, the loudspeakers can destroy the apparent integration by a too-wide a response. If some of the sound for a particular program component, say the triangle or something with considerable high frequency in it, is radiated by more than one loudspeaker, unrealistic loss of "identity" can result. Even though one unit reproduces stronger than the other one, the distribution of these extreme high frequencies, with their much shorter wavelengths, is bound to be extremely different from the original. This is because the interaction pattern comes from two sources instead of just one. For this purpose a single loudspeaker unit with a wide-frequency range on single-channel could sound more realistic than the two loudspeaker units connected to stereophonic program.

Consequently it may sometimes even be advantageous not to have quite such a wide frequency range when using stereophonic



Courtesy Jensen Mfg. Company.

Fig. 46B. The Duax: A further example of design in extended range loudspeakers.

presentation, to avoid the possible confusion effects of the extreme high frequencies.

Restriction of the frequency range in this way has another advantage which many high-fidelity listeners will appreciate. That is the reduction of effective background noise. It has long been commented that the only thing that super-tweeters give, to some program material at least, is some additional "hiss". In many instances this is partially due to deficient design of the tweeter, because a unit with a frequency response consisting of a sequence of resonances or peaks, serves to accentuate background hiss far more than it improves the realism of the musical or program presentation. For this reason the impression often conveyed is that the tweeter adds nothing but hiss to the presentation.

With a good tweeter having a level and smooth frequency response, this is not so true. Background hiss is not apparently increased out of proportion in this way, and there is improved accuracy in the presentation of the different program components. However, background noise from various sources can still increase by the extension of frequency range at the high end. This is because background noise, in the form of hiss, is so much energy *per cycle*. In other words, there is as much hiss between 7000 and 14,000 cycles as there is all the way from zero to 7000 cycles. In terms of program, this means that extending the frequency response by just one octave, in a region where all that is gained is improved accuracy of sound already heard, doubles the background noise.

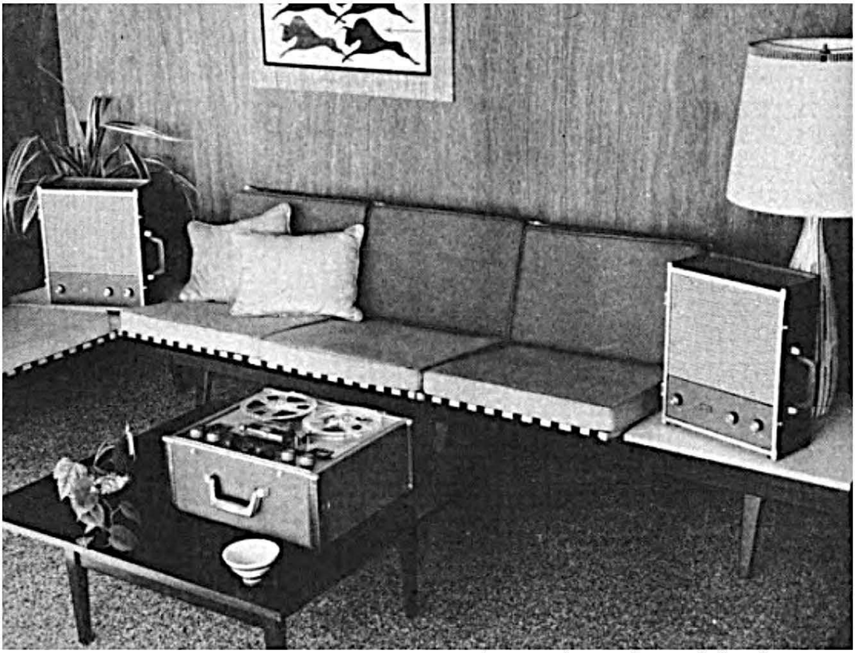
Use of stereophonic presentation with a response that decreases somewhere after about 7000 cycles, will achieve the same results as regards accuracy of presentation, without the undue increase in background noise.

Placement

The net result, then, is that stereophonic presentation enables us to make do with a cheaper variety of loudspeaker. In fact a loudspeaker with a more restricted frequency range is, generally speaking, an advantage for stereophonic presentation rather than otherwise. Also affecting the relative noise in some systems of stereophonic is the fact that using more channel in the same space increases the natural background noise of the recording medium. For this reason it is advantageous to restrict the playback range to come out with a respectable apparent background noise and a useful overall dynamic range.

The next question that is going arise in loudspeakers for stereophonic presentation is "How should they be placed?" Some systems and some manufacturers recommend distributing these loudspeakers along one wall of the room. If two loudspeakers are used they should be either in two adjacent corners of the room, or a little way in from the two corners along the wall (Fig. 47). If the system is three-way then an additional speaker is required in the center. (It's surprising how often this proves impractical because of conventional layout or furnishing of American living rooms!).

Another school of thought prefers to have the loudspeakers in the same cabinet facing out in divergent directions, so as to achieve



Courtesy Ampex Corp. of America.

Fig. 47. Typical two-way separate loudspeaker placement, demonstrated with a portable stereo system: Model A-122.

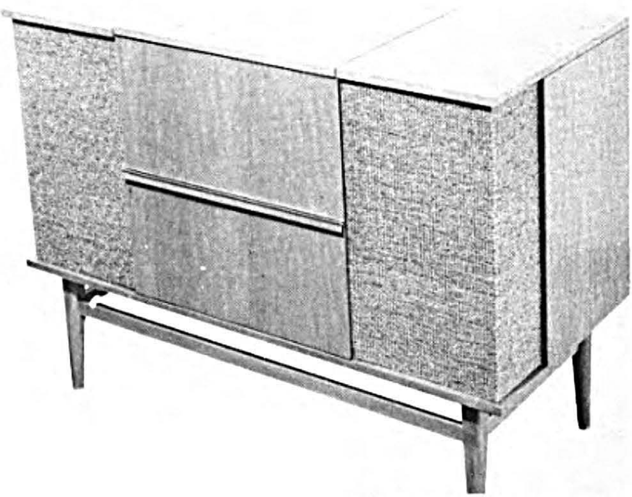
distribution of the sound pattern radiated in this way (Fig. 48). The question one most often hears is, "Which is the best?"

The answer to this question is by no means a simple one. We will need to devote a little consideration to it from the viewpoint of the different factors that influence the decision. This is partly dependent on the system of stereophonic presentation used — how the recording is made and how the various channels differ from one another — and partly on the arrangement of the listening room.

According to System

To avoid getting too confused, let's consider each of these items separately, although this is not entirely possible, because the two are to some extent interdependent.

Taking the two-channel stereophonic first, the ideal loudspeaker placement depends on the way the recording is made. If the recording is basically binaural, which means the two microphones



Courtesy Ampex Corp. of America.

Fig. 48. A unitized stereo system with the loudspeakers divergent from the same cabinet. Tape, phono and radio are housed in the center section: Model A-423.

are spaced apart by approximately the same distance as the ears in the human head, in a similar and suitable obstacle, there is usually an advantage in having the loudspeakers as far apart as possible, even though some have dubbed this "bistereonauralphonic".

This is because we want to ensure that each ear of the listener hears *principally* the program intended for that ear. The ideal listening position would be in the center of the room with loudspeakers disposed on either side of the listener. They could be on opposite sides of him, feeding toward each ear (Fig. 49). But for this to be successful the wall space behind each loudspeaker would need to be acoustically damped with heavy curtaining and acoustic tiles, and possibly some obstacles to break up the sound wave. This is to prevent undesired reflections from each loudspeaker back across the room.

For this reason, in most rooms, the best arrangement is to have the two loudspeakers occupying adjacent corners with the listener midway between the two. If the listener is a little nearer one loudspeaker than the other this slightly invalidates the impression conveyed from a binaural recording, but it still sounds better than a single-channel presentation.

The fact that the phase difference on the original recording represents only the time between the two ears on the dummy head, means we can allow a greater phase difference in the listening room without getting into confusion troubles. Consequently the listener can sit nearer to one loudspeaker than the other without experiencing serious confusion.

If the two-channel recording is of the type where the microphones are more widely spaced, then listening in a position equally

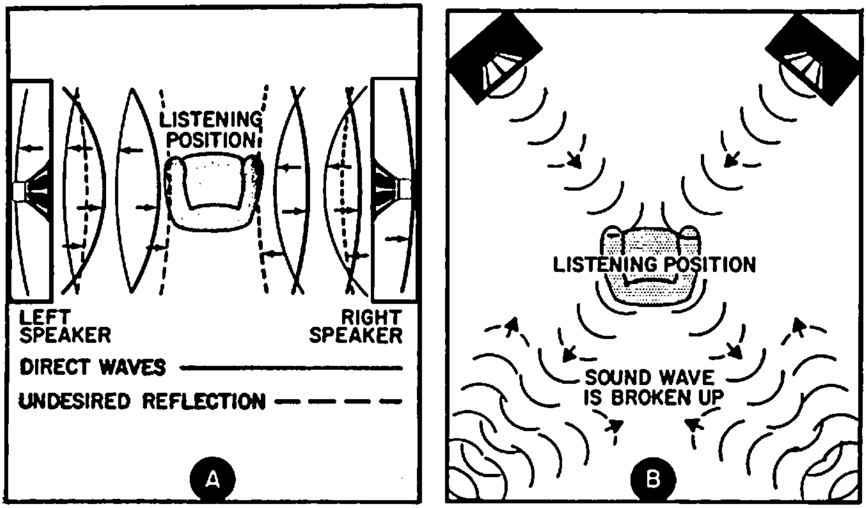


Fig. 49. Positions for listening to so-called binaural recording: (A) directs sound most specifically at each ear, but produces undesirable reflections: (B) a better practical compromise avoiding most reflections.

distant from the two loudspeakers, with the loudspeakers located in adjacent corners, may still give a successful illusion. But in this case the phase difference already existing in the two channels is greater than the natural phase difference between ears. Consequently any additional time or phase differences, occasioned by the listener moving from the position of equal distance, makes the sound become somewhat confused. This would be particularly noticeable in the type of program that contains a rapid succession of short duration transients, such as strings being played *pizzicato*.

As most people like to listen with their friends rather than just by themselves, we need a system that gives a more effective presentation to people listening in different positions around the room. This is why the type of loudspeaker cabinet in which both loudspeakers are contained in the same enclosure, facing outwards from opposite ends, has developed (Fig. 50).

If the loudspeakers were brought together more closely, side by side, the time and intensity differential that provides the stereophonic effect would be reduced. The effect would be very similar to mixing the program content of the two channels and then play-

Courtesy Bozak Mfg. Company.

Fig. 50. Another variant of the unitized loudspeaker system, a provincial version of the cabinet shown in Fig. 28.

ing them over one loudspeaker, because the loudspeakers are so close together that the sound waves would merge although the electrical impulses driving the loudspeakers would not. Placing the loudspeakers back to back, however, obviates this difficulty by having the sound waves radiate in opposite directions. But because they start from *sources* that are still close together, the time difference does not become objectionable in any part of the room, as it can when the loudspeakers are widely spaced.

Here we could give a little attention to how different frequencies are radiated. At low frequencies, the sound waves from the two channels are virtually in phase, because the wavelengths are big compared to the distance between the original microphones. Similarly, in the listening room, usually the low frequencies are such that only about one wave — or perhaps only even a fraction of a wave — can be contained in the room. This means that both loudspeakers will be aiding one another. The diaphragms will both be pushing at the same time, and both pulling at the same time. So it does not matter whether the loudspeakers are close together, far apart, or even on opposite sides of the room. They both augment the total sound wave at that frequency, and there is no appreciable sense of direction to these frequencies.

Where the loudspeakers are mounted in the same cabinet, if the bass reflex principle is used, a common port can be provided with dimensions suitable for two loudspeaker units and the effects will be quite good. At the middle and higher frequencies, the absorption inside the cabinet will prevent interaction troubles and each loudspeaker will radiate its own sound independently from the front of its diaphragm.

Placing the loudspeakers in opposite directions from approximately the center of one wall of the listening room, also minimizes interference due to the extreme high frequencies, to the extent that they are radiated. If a person is sitting so he hears one loudspeaker more or less directly, he will hear practically nothing of the extreme high frequencies radiated by the other unit to give him the confusion effects noted earlier. If a listener is located in front of the cabinet so he is not directly on axis of either loudspeaker, the extreme high frequencies will not reach him very strongly at all. They may bounce off the end walls and reach him at approximately the same time from both units, but they will be considerably attenuated (Fig. 51).

In doing so, however, because he is in the best position for enjoying the stereophonic effect, he is least in need of the high frequencies to improve his sense of realism. Thus this arrangement gives the best all-round performance on two-channel stereo of conventional type.

Three-channel stereo for home use is not quite such a simple matter to decide. With the single assembly, placing the side speakers facing outwards at each end and putting the center one on the front facing the room, a possible method is to operate the front speaker at a little bit lower volume level, so the correct effect is achieved when sitting right in front of the cabinet. This arrangement has the advantage of being most tolerant of phase and intensity differences, whether you want to listen right in front or in some other part of the room.

As with the two-channel system, the remarks about spacing also apply to three-channel. If the recording is made from three widely-spaced microphones in the studio there will be a considerable phase or time difference, amounting to several waves of higher frequencies. If this is played back over three loudspeakers equally widely spaced, the effects can be somewhat confusing because of the time difference. It is better under these conditions to use the single integrated loudspeaker with one speaker in front and one on each end of the cabinet.

But sometimes the three-channel stereophonic program is synthesized from recordings of separate groups of the composite audio, or sometimes it is produced in a manner very similar to the single-channel coded – each of the three channels is produced by varying its amplitude from a common composite audio, and recording that on each channel. In either case, the program contents of each channel is identical at every instant, but the intensity of the program on the different channels varies according to the sense of direction intended.

If either of these methods is used in making the original recording, the playback characteristics will be very similar to coded stereophonic, and quite a satisfactory impression can generally be obtained by using separate loudspeakers placed apart along the wall, preferably in a room that is rather oblong than square, perhaps with the length one and a half times the width. It is then best to arrange the loudspeakers across one of the shorter walls of the room and listen toward the other end. This gives a greater listening area where the sound will be heard in the correct perspective intended by the system.

Where the relative sense of position is obtained essentially by intensity differences rather than the more usual combination of intensity and phase differences, in which case there is as a rule a bigger difference in time than intensity, the best arrangement is to use the loudspeakers fairly widely separated, because this utilizes the basic principles on which the stereosonic system is based. For stereosonic presentation speakers spaced apart along the wall are by far the best arrangement, because this gives the difference in intensity from each loudspeaker the best opportunity to produce the relative sense of direction at the two ears of the listener, wherever he may be in the room.

So our conclusion about loudspeaker type and positioning is quite similar to the one we drew when discussing single-channel sound. In that case we found that one loudspeaker might be good for presenting solo performances while another one sounds better for full orchestra, and so on. In this case we have other differences to contend with, so the positioning and arrangement of the loudspeakers to give the best stereophonic presentation, depends on

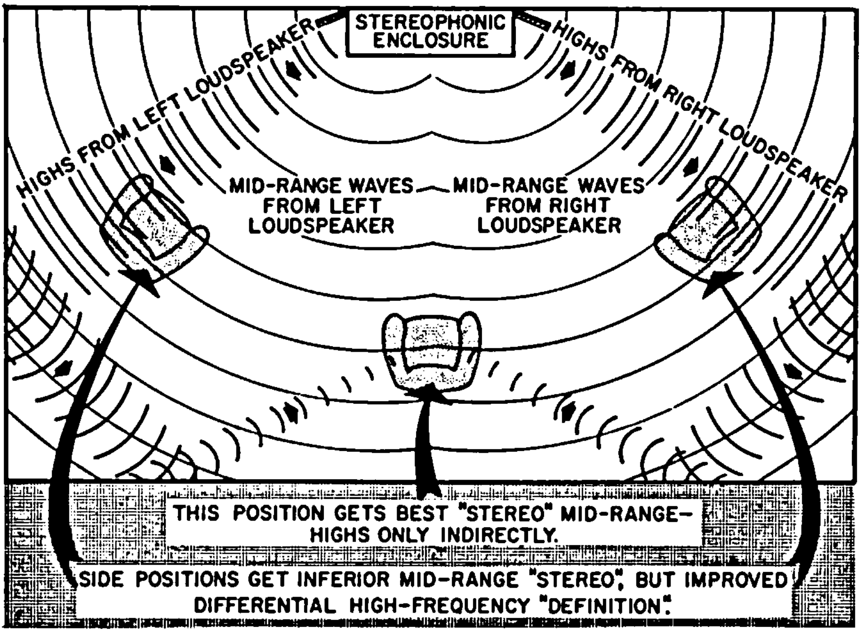


Fig. 51. Realism in different parts of the room is due to different effects, using the divergent or outward-facing arrangements of Figs. 48 and 50.

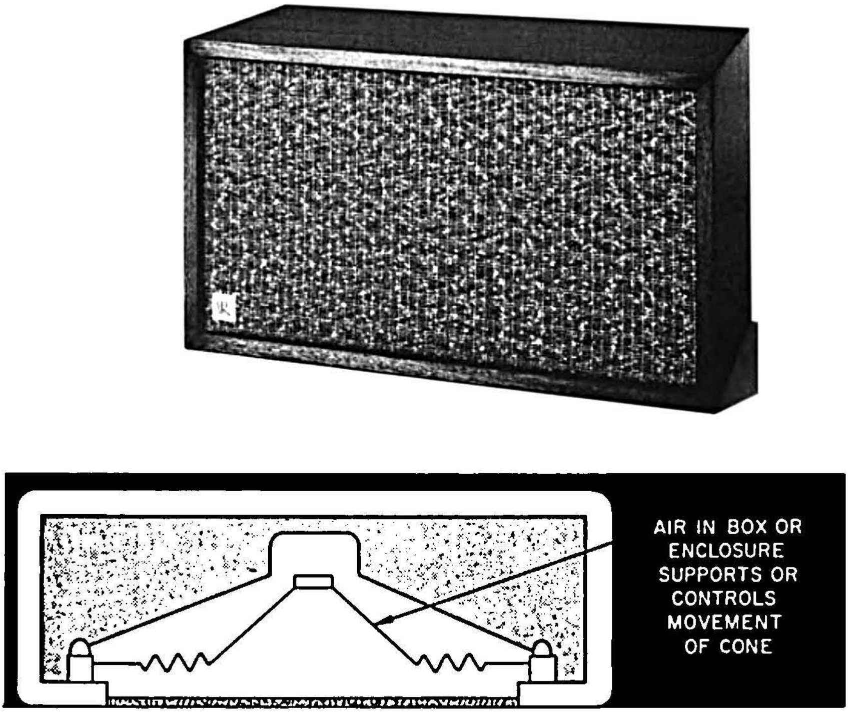
the way the program was recorded in the first place – how the mikes were placed or whether the program was synthesized. No one arrangement proves to be ideal for all forms of presentation.

According to Listening Room

The size and shape, and to some extent the form of decoration in the room, also affects the choice of loudspeaker. In describing loudspeakers for single-channel high-fidelity presentation, it has been suggested that it is good to have the loudspeaker somewhat commensurate in size with the room in which it will be played. A large elaborate loudspeaker in a relatively small room makes the sound seem confused. On the other hand a small loudspeaker unit, even though it may produce adequate power for the larger room, will seem to be lost because the sound source seems too small.

This principle extends into the use of loudspeakers for stereophonic purposes, to an even greater extent. At the beginning of this chapter we pointed out that in general the large elaborate loudspeaker was not suitable for livingroom presentations, but that it might be quite suitable for use in a large auditorium.

Extending this principle downwards in size, the speakers used for stereophonic presentation in a relatively small livingroom should be quite small units. As each of the stereophonic units will be working in phase for the lowest frequencies, and because the room is probably small enough so that the lowest frequencies only contain part of a wave, it is quite possible for the smaller loudspeaker



Courtesy Acoustic Research.

Fig. 52. The Model AR-2. Top: Enclosure and loudspeaker system. Lower: The acoustic-suspension principle used for the woofer unit.

units — that would be regarded as inadequate for bass response in a single-channel high-fidelity system — to prove quite effective when used for stereophonic presentation.

The small, acoustic-suspension type loudspeaker (Fig. 52) can prove very effective for stereophonic presentation in small rooms. It is somewhat inefficient, and will probably need a 25 to 30-watt amplifier to drive each loudspeaker to allow a sufficient margin to handle the higher transient peaks in the program. For this reason a somewhat more efficient unit is the miniaturized reflex-type, which uses an escape from the rear of the diaphragm to produce an in-phase component at the front (Fig. 53). For single-channel presentation these might appear to be somewhat bass-deficient. But in a small room they can sound very good, and when two or three are used, as the case may be, for two-channel or three-channel stereophonic presentation, the bass can prove to be quite adequate.

Stereophonic presentation really begins to sound better, though, in the medium to larger size livingroom, because it gives the sound an opportunity to spread better. In the larger size rooms it is well to have a somewhat larger type loudspeaker. If cost is a serious factor, an extended range, maybe with a low-cost tweeter mounted close to it, will give very satisfactory results.

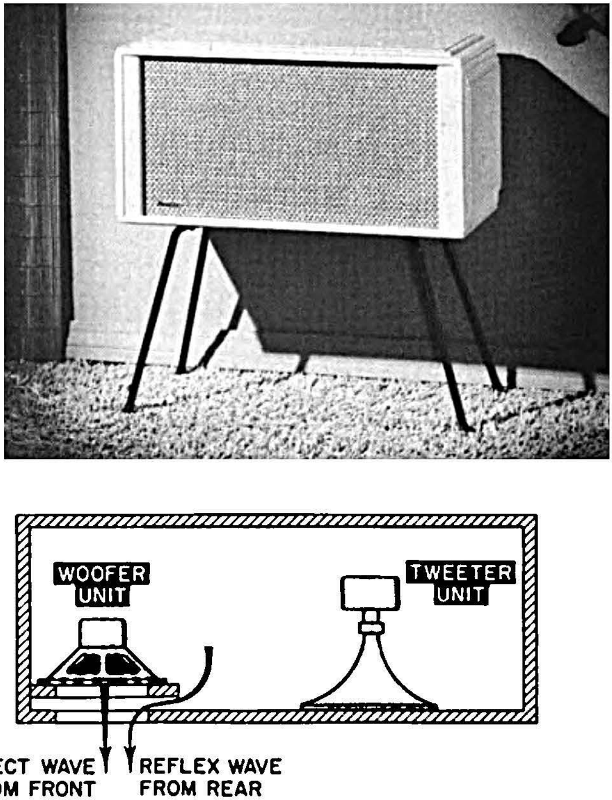
It is not generally necessary to go in for a super-tweeter with stereophonic presentation, because these "super-high" frequencies add very little when stereophonic presentation is available. The increased accuracy in high note rendition is achieved by other means. However, for the two-channel type where the two loudspeakers are mounted on opposite ends of the same cabinet, which is also suitable for some of the smaller rooms, tweeters and possibly super-tweeters can be an asset, because they improve the impression of realism when listening from a position toward one end rather than in the middle-front. In the middle-front position the tweeter top end and response and the super tweeter are probably not audible at all, due to absorption around the room. But here the stereophonic effects get the full benefit and consequently the super-tweeter is not needed.

So much for the loudspeaker part of the question. There are other components, however, that are needed for the different systems, so we will give a little attention to this matter here, to clarify just what is involved in using the different systems and what extra equipment is needed for each.

The Complete System

It goes almost without saying that any stereophonic system will need two or three loudspeaker systems, as the case may be, and a corresponding number of power amplifiers to drive the loudspeaker systems. The loudspeakers may be separate or in the same cabinet as we have already discussed. Similarly the amplifiers may be completely separate units or may be powered from a common power supply. This may effect some economy provided the power rating is such that a convenient power supply can be designed that does not require just twice as many components as the supply for only one of the amplifiers.

A possibility with a two-channel system that some manufacturer may some day exploit, is the use of a two-channel power amplifier



Courtesy Jensen Mfg. Company.

Fig. 53. Treasure Chest Duette: Top: A useful type loudspeaker system with miniaturized bass reflex arrangement. Lower: Sketch shows basic layout.

that uses a single-unit double-output transformer. This would utilize a single magnetic core with two separate transformers wound on the same core (Fig. 54). There will of course be a degree of coupling at the low frequencies, but the high frequencies will be adequately separated.

It might even be possible to effect tube economy by using a large power-rating output tube single-ended, instead of using push-pull output for both channels. In this way the distortion at the lower frequencies, which is where the curvature usually produces most intermodulation, will be neutralized by the fact that both channels are approximately in phase to these frequencies. The amplifiers would of course be arranged so that one output tube pushes when

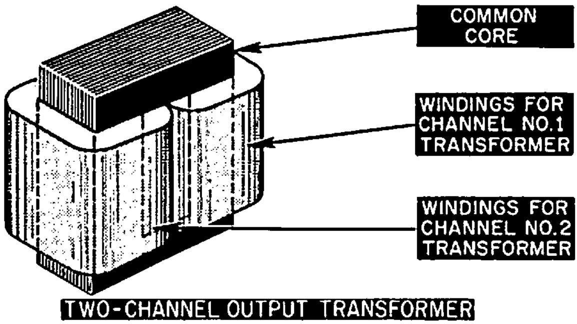


Fig. 54. A possible economy in amplifier design would be achieved by using single-ended amplifiers for each channel, with a combined two-channel output transformer, illustrated here.

the other pulls, so at extreme low frequencies the amplifier behaves just like a normal push-pull amplifier.

Some people may be bothered about the interaction that can occur electrically between two amplifiers that share an output transformer in this way, for economy purposes. Actually the interaction will be no greater than the acoustic interaction that occurs when the two loudspeaker systems for a stereophonic system are mounted in the same enclosure at opposite ends. So if you are going for the composite-speaker arrangement for your stereophonic system, there would be absolutely no objection to utilizing an amplifier of this type, if and when such an amplifier appears.

Figure 55 shows an amplifier that has been engineered to give economical design by putting two amplifiers on one chassis, with simplicity of control, by combining both into single-knob operation, with a balance (called "focus") to equalize channels if necessary.

For the rest of the components, the two- or three-channel stereophonic requires two or three preamplifier channels complete, together with separate radio tuners if it is intended to pick up program from the air. Most stereophonic program will probably come from the two-channel or three-channel tapes, so separate tape heads are required, or a composite head with three separate pickup arrangements combined in one (which will cost almost as much as three separate heads). The individual channel amplifiers will need to be just as good as for a single-channel high-fidelity system.

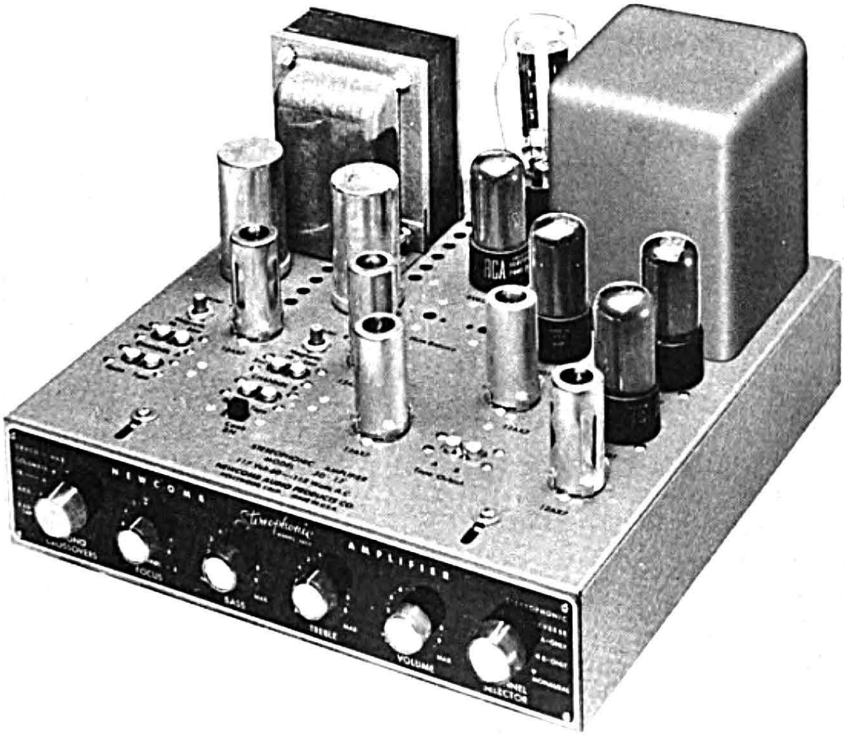
The stereosonic system again needs both two-channel pickup arrangements and two-channel preamplifiers. If it is used with the number 1 high-fidelity, number 2 low-fidelity channel combination, it may need a recombining arrangement to obtain the necessary program channels for the two loudspeakers. However this is quite a simple and inexpensive addition to the system, and would require only one extra tube and a few small associated components. If stereosonic is used with the single-groove disc recording, a special pickup will be needed that will be somewhat more expensive than a regular pickup, but can probably be produced for less than twice the cost of the better class pickups.

The coded-stereo system avoids the need for multichannel pickup or playback head, together with preamplifiers or radio channels, but instead requires a unit which, for the theater system, has been called an "integrator". This is a unit which separates off the control frequencies and then provides separate controlled outputs which vary in level in accordance with the dictation from the control frequencies. For the theater systems, this unit by itself costs in the region of $500, but it utilizes expensive filtering arrangements and components of a quality to conform with general theater and professional practice. It is quite possible that a coded stereophonic system will be developed with filters that perform a very similar purpose quite effectively for much lower cost.

The system cost for a coded-stereo three-channel arrangement will probably be quite comparable with a two- or three-channel system for regular stereo. The big difference is that the cost of recorded programs, if this system becomes used, will be in the region of half that for regular stereo, because only one track is needed. However until such a system is developed this discussion is somewhat hypothetical.

Building a System

The last question we need to discuss in this chapter is how the reader may acquire a stereophonic system. Some will already have a high-fidelity system they may have used for several years, that includes a good power amplifier and loudspeaker system. From the foregoing discussion of ideal loudspeaker systems to use, it may appear that the single unit they have is not ideally suited for stereophonic reproduction. So should they throw out the present sys-



Courtesy Newcomb Audio Products.

Fig. 55. An example of economic design in a complete two-channel amplifier for stereophonic presentation from disc, radio or tape: Model 3D-12.

tem completely and acquire a completely new stereophonic system? This certainly will give them, ultimately, the best results. However, if they still wish to play their single-channel high-fidelity program material, undoubtedly their present system will sound better than trying to play single-channel over their new stereophonic system. So it may be advantageous to retain the high-quality, large loudspeaker system they already have.

If they wish to avoid the high immediate expense of a complete stereophonic system, it is quite feasible to use the present high-fidelity system with one channel of the new proposed stereophonic system to supplement it. This will not give quite such good stereophonic realism as two identical speaker units, but it will certainly give a very credible presentation of stereophonic program material. This will enable the user to attain his program improvement in

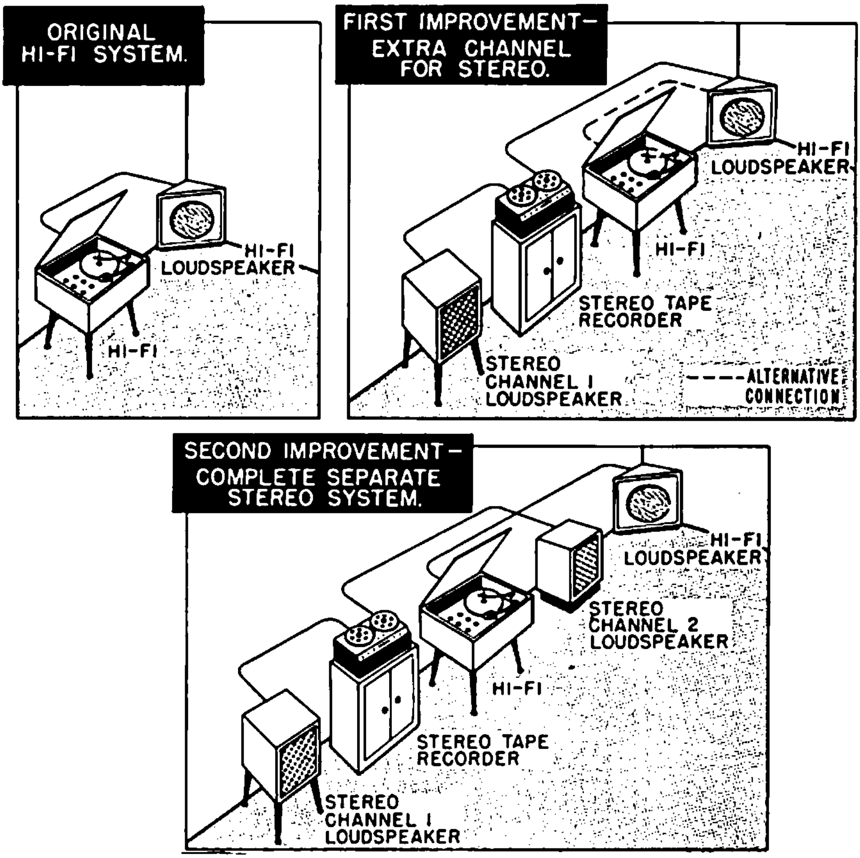
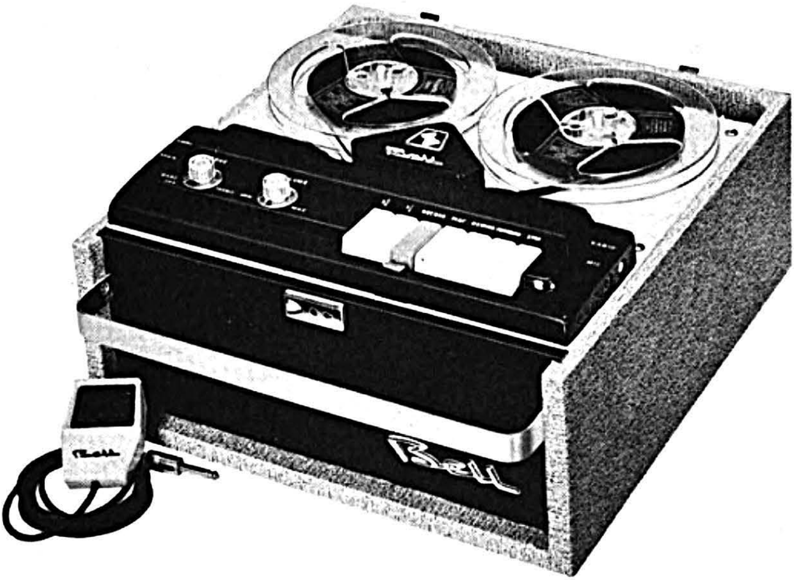


Fig. 56. A progressive program for changing over to stereo — the improvement and cost are both distributed in this way.

steps, as he can afford to buy the different stages. Later on he can buy a second channel identical with his first, and then he has a complete stereophonic system (Fig. 56).

Those buying high-fidelity systems for the first time may want to buy for single-channel with a view to later expansion into stereophonic. At present the amount of stereophonic material around is considerably less than the quantity of single-channel recording, so there are still advantages in using single-channel because a greater variety of program material is available. This being the case, for some time yet prospective high-fidelity customers will want to buy single-channel equipment and expand later on into stereophonics.



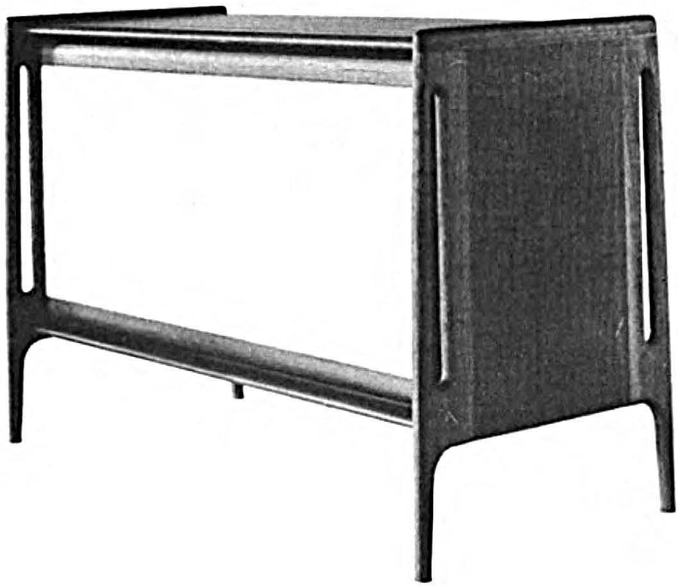
Courtesy Bell Sound Systems.

Fig. 57. One of the recent stereophonic tape-recorders that will play two-channel tapes and record single channel.

If this is the way you are starting, two loudspeakers certainly do sound better than one, so it will be good to invest at the outset in the stereophonic loudspeaker system you ultimately intend to use. However, until you can afford the complete stereophonic system, you can make do with a single-channel disc playback arrangement, using a single-channel preamplifier and power-amplifier to feed the two separate loudspeakers connected in parallel. As two loudspeakers, to give ultimate stereophonic reproduction comparable with a single, high-quality loudspeaker, will not be very much greater in cost, this seems to be the most economical way of approaching such an expansion program.

Using this method, you only need to buy one of the amplifier systems at a time. Later on, when you get into stereophonic presentation, you can buy the two-channel tape-recorder. Some of these provide preamplifiers for both channels, but a power-amplifier for only one. This unit will then supply all the extra components you need to convert your single-channel system into a stereophonic system.

Many of these new two-channel stereophonic tape-machines are



Courtesy Integrand Corp.

Fig. 58. The "integrated" loudspeaker system shown here contains two complete three-way loudspeakers, each with its own servo-drive transistorized amplifier. The units for each channel are mounted behind the grill-cloth at a divergent angle.

not two-channel *recorders* at all. They are primarily designed as two-channel *playback* machines for playing stereophonic tape. This is because most people will not have the facility for making their own stereophonic tapes. They may, however, wish to make tapes of things they do at home, in which case they will use the recorder as a single-channel recorder. Facility is usually provided on stereophonic tape-machines for single-channel recording only (Fig. 57).

In some cases the record arrangement is on one track only, in which case the tape can be used as double-length in the same way as any half-track recorder, playing one side and then flipping the reel and playing the other. In playback, only the side used for recording will be used. The other playback head will be disconnected or turned off.

In other cases the record head of a stereophonic tape-machine is full-track, so that it records all the way across the tape. This enables playback to be made through the stereophonic system as a single-channel arrangement. Both channels will get exactly the same sound because the two pickup heads will be traversing the same recorded track. This method, of course, will only work with the stacked head arrangement, not with the staggered.

A very recent development that makes a good starting point for a stereo system is a servo loudspeaker system, in which two complete three-way, well-integrated loudspeaker systems are mounted in the same enclosure, and provided with their own drive-amplifiers — transistor operated — and crossovers, ahead of the amplifiers. This can conveniently follow a single-channel "front-end" — radio, phono or tape — and later, without any addition to the loudspeaker at all, can be fed stereo program (Fig. 58).

Chapter 6

STEREOPHONIC SYSTEMS FOR MOVIE THEATERS AND AUDITORIUMS

Most people interested in stereophonic reproduction are primarily interested in what is needed to give them this kind of reproduction in their livingroom. But as soon as you get interested in a subject, anything connected with this subject attracts your attention. So most stereophonic sound enthusiasts will also be interested to know "how it is done" in the movie business. This might be regarded as just satisfying idle curiosity, but it also serves a useful purpose. Understanding the different problems associated with sound reproduction in *large* places, such as movie theaters and auditoriums, helps to understand the opposite problem of reproducing sound in small spaces, such as the livingroom. The problems often encountered are completely reversed.

Techniques

For example a presentation in the movie theater may be intended to convey the impression of a livingroom scene, together with the appropriate sound effects, but the actual area in which the audience is listening is a large auditorium. In the home reproduction systems, the program desired may be a concert from a large auditorium, but the listening place is a small livingroom.

Of the two problems it would seem that presenting an impression of a concert hall in your livingroom is the easier of the two. The reverberation can be recorded along with the program and this is reproduced in the livingroom to give the effect of the larger auditorium. In the opposite direction, the lack of reverberation can be recorded on the sound track corresponding to the small room, but you cannot take reverberation out of the larger auditorium in which you listen. The best, in fact the only approach, is to deaden the auditorium. This does minimize its reverberation, so that after sitting there for a short time you are no longer conscious of it, and the sound you are conscious of is that reproduced over the system without the comparatively small degree of reverberation added by the auditorium.

The important difference between the two kinds of system is the fact that the listener sits at a greater distance from the loudspeakers in the movie theater (except possibly those who sit in the front rows). The average listener is at a distance that is several wavelengths, even of the lowest frequency, from the loudspeaker grouping. Consequently *some* of the things that may be said about stereophonic sound presentation in an idealistic sense, and which do not apply fully in the livingroom because of the limited dimensions, do apply, to a greater extent, in the movie theater.

On the other hand, the spacing between loudspeakers, even the three units generally employed behind the screen, is so great that it is impossible to base the presentation on *phase* difference effects, because the space in between loudspeakers represents several wavelengths of all except the very lowest frequency sounds. Consequently the most satisfactory method of obtaining a suitable stereophonic sound track presentation in a theater consists of using separate audio channels and synthesizing the individual tracks to represent the different program content.

A multiple-track recording of an orchestra, for example, will be made from microphones close to separate groups of the orchestra, so there is very little time difference between the different sections of the orchestra on the original separate channels recorded. Then, to get the desired positional effects in the auditorium, to correspond with the visual presentation of the orchestra, the different original channels which may be as many as six or more, are mixed into the final stereo tracks to get the required directional effect with each of the different components originally recorded separately.

In some theater systems, "surround" loudspeakers are used in addition to the three loudspeakers behind the screen. These are fed from a fourth sound track and can contain either reverberation material to give the effect of echo, or else can contain completely separate effects such as a choir singing behind the audience. Full utilization of these facilities is not as effective as might be anticipated, because of the big time differences required for sound to travel to different parts of the auditorium. What may be an effective presentation for one seat would not be in another, due to the different distances from the loudspeakers feeding the different positions (Fig. 59). For this reason program material has to be chosen very carefully so the time difference in the receiving of program from different directions is not critically important to the impression conveyed.

For example, a composite choir could be presented over the surround loudspeakers while maybe a solo comes from the stage, thus avoiding any specific time problem. If the arrangement were intended to give an echo effect, the echo might be unduly long from some locations in the theater and almost non-existent in others.

Due to this limitation in the possible application of this kind of effect, what at first appeared to be a wonderful prospect has practically died out. Movie producers hardly ever use surround effects any more.

Systems

The actual systems employed in movie presentations are very numerous. There are so many different wide-screen techniques that it would take a book on its own to describe the picture presentations. Along with the different forms of picture presentation, we have corresponding variations in the sound presentation. But, from the viewpoint of stereophonic recording, we can divide them under three principle headings. Other differences are mainly concerned with where the sound tracks are located on the film relative to the picture, and how many there are.

1. Multiple-Track Optical

The earliest attempts at stereophonic presentation in the theater were made before the advent of magnetic recording, or at least before the modern adaptation of it by means of magnetic oxides on an acetate base. Consequently the only way to provide stereophonic effects was to use extra sound tracks of the type already in use.

An advantage of the optical sound track is its good dynamic range, because of the noise reduction facility. Background noise had always been a problem with movie sound tracks and this resulted in the development of noise-reduction systems. This is an arrangement which cuts down the amount of light passed through the film in proportion to the sound level. When there is only a very feeble sound track it is also a very narrow one, and the rest of the track is completely opaque. As the sound becomes louder, the width of the track is opened up allowing a greater amount of

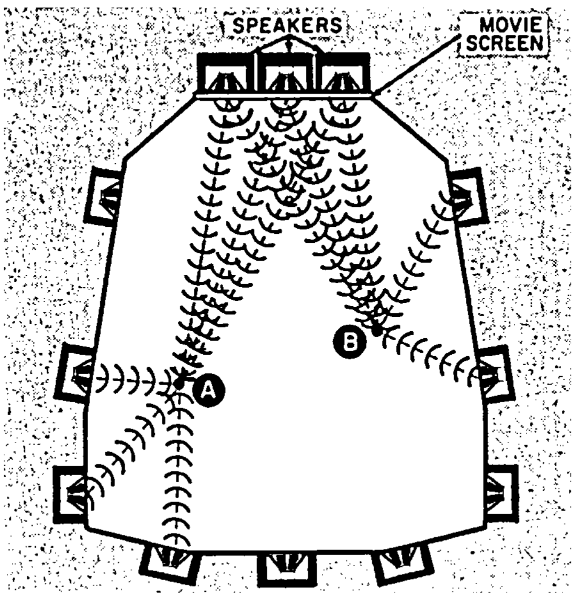


Fig. 59. The problem with the use of surround speakers. Listeners in different positions (A and B) get sounds from the screen loudspeakers (heavy dashed lines) at different intervals from the sound from surround loudspeakers (light dashed lines). The listener at A will hear the surround speakers considerably ahead of the screen units.

light to penetrate, which is controlled in turn by the variations in density or width of the clear section of the film (Fig. 60).

The exhibitor's problems with multiple-track optical presentation are very considerable. The movie projectors installed in the average theater have the sound heads located in the correct position for standard optical track. To present stereophonic optical, which has additional sound tracks located somewhere else on the film, requires the addition of extra sound heads in the projector. Usually this means the complete installation of a new section of the projector to replace the old sound head section, a very expensive proposition.

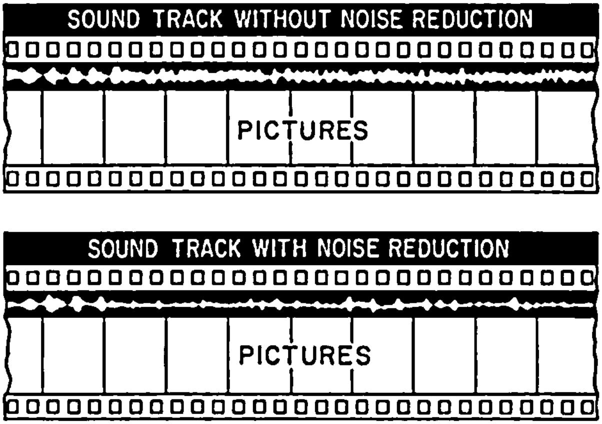


Fig. 60. How noise reduction works on optical sound track ("variable area"). Quiet sections are narrowed to limit light transmission and reduce background noise.

2. *Multiple-Track Magnetic*

The advent of magnetic tape-recording made possible the application of the magnetic oxide to the actual movie film. This is done after the printing of the picture on to the film by a process called "striping". Then the sound is "printed" on the stripe, not by a regular printing process, but by a magnetic-recording process. By analogy with the process of producing the picture and the older optical tracks, the word "printing" has stuck, althought it is difficult to see how it applies in any literal sense.

Magnetic striping has its problems. It is very difficult to get one reliable magnetic stripe on to the film, let alone three or four. They are apt to be insecurely cemented to the film so that pieces flake off. If they don't flake off, they wear off relatively easily, because they are so narrow compared to the regular ¼-inch tape used for proper tape-recording.

Due to the narrowness of the sound track the dynamic range is not very good at best. It can sound deceptively good in a stereophonic system by keeping the volume level pretty constant. No passages are allowed to drop down to a very low "intimate" level of presentation. This, of course, is not too much of a burden on many movie producers, because they rather like to have the intimate "whispers" on the screen loud enough to almost deafen the audience!

The compatibility problem here is solved more readily, because the magnetic section of the projector clamps on the top and is often called a "penthouse" for this reason. In this position it will not interfere with the regular functioning of the projector on optical sound films, but means a changeover is necessary, by means of switching, to transfer from single-channel optical, to multichannel magnetic.

When the film is new and the oxide tracks are in good shape, and the playback heads in the penthouse are brand new, the quality is very good. Unfortunately this quality does not maintain too well. The heads wear down, as well as the oxide wearing off the film, so that quality deteriorates both as a system and also with usage of a film. This is a defect that occurs to a much lesser extent with optical track.

The manufacturers recommend replacement of the penthouse heads at regular intervals, but this is an extra chore and one that is apt to get overlooked. All the while the system still plays, no change is made, consequently many of the magnetic stereophonic systems continue operating in very inferior conditions.

3. *Perspecta Sound*

This uses a single-track coded presentation, which we have discussed earlier in the book. It could easily use either magnetic or optical sound track as a base, but it was developed in optical. This made the system completely compatible with single-channel presentation.

The system even includes a relay that automatically switches from single-channel presentation over the center-channel loudspeaker to three-channel presentation over the three separate loudspeaker channels when the Perspecta coding signal comes on. This relieves the projectionists of all responsibility for changing over connections according to what kind of film is running. This is not meant to suggest that projectionists are lazy. But when time comes to change from one reel to the next, he may have so many things to do at once that one can easily get overlooked. It is even possible to splice sections of film having single-channel and Perspecta sound tracks into the same reel, and the system will automatically switch over from one presentation to the other as soon as the sound track changes.

Of course, the Perspecta system has the same limitations mentioned with the coded system earlier in the book. It cannot give simultaneous direction identity. But it is surprising how realistic this presentation is, especially in view of the fact that in theater presentation all sense of direction is more dependent upon relative intensity from the different channels than it is upon phase differences. As the coded system depends entirely upon intensity variations, this enables it to exploit this difference to at least as good advantage as the regular multichannel stereo.

An advantage of the coded stereo for movie presentation is the fact that the system itself enables further noise reduction to be effective by means of the coding frequencies. The three channels can be turned down to lower level, or one channel can be turned down and the other channel turned right off, or any combination desired. But the effect is that background noise is turned down along with the program sound.

This amounts to electronic noise reduction, which is additional to the noise reduction that can already be incorporated on the optical sound track to minimize noise that may occur due to dust or other particles on the surface of the film. Applied to magnetic recording (which at the date of writing has not been done) this process could give magnetic recording the advantage of noise reduction and greatly improved dynamic range.

Making Movie Sound Tracks

So much for the systems used for film and theater stereophonic presentation. Regardless of which system is used to carry the stereophonic program material, whether it is a magnetic system or optical and whether the system uses separate tracks or a coded arrangement, the method of work for producing the ultimate program that will be heard from the film is very similar at the studio, and quite different from that generally used in producing program material for home presentation. This again reflects the difference between the requirements for home and theater presentation of stereophonic sound.

Film studios usually make at least six separate master sound tracks from which the ultimate composite is made, to be presented on the film (Fig. 61). This is true even for the older nonstereophonic presentation. The background music is usually made in about three separate tracks that contain different groupings of the composite orchestra. This enables the complete program to be recorded, and different parts of the orchestra can be emphasized or toned down in producing the ultimate composite, according to the "feeling" the director desires to be associated with the pictorial presentation it happens to accompany.

If the pictorial presentation shows a person singing a vocal solo,

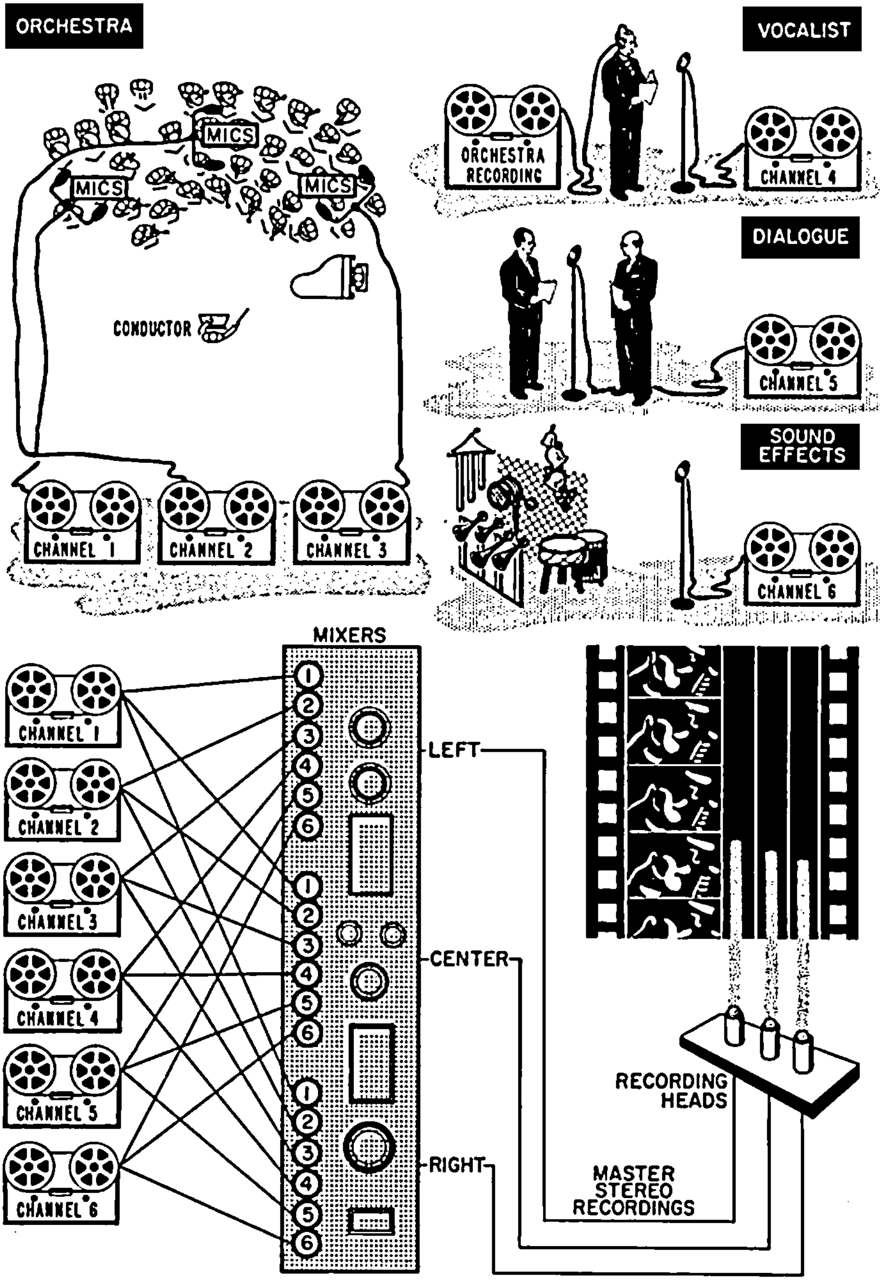


Fig. 61. Multiple channel recordings are used to synthesize motion picture sound tracks, using "panning" mixers to get the right mixtures into the three master channels for the final "print".

this will require another track. This is recorded completely on a separate track so as to allow of adjustment being made in producing the final composite, to match up with the changes in viewpoint presented by the camera.

For example, when the camera shows the soloist front view closeup, the sound track could emphasize the solo and put all the other music into the background. If the shot changes to a picture of a section of the "audience" listening to the soloist, then the sound needs to change to give this effect at the same time. By having the complete program recorded on separate tracks this change can be produced in making the final composite to go with the picture.

Another track is required for dialogue. Dialogue may or may not be accompanied by background music (or perhaps the sound of a soloist continuing with a performance, all going on at once) so we need another track for the dialogue that runs right through the picture.

Finally, all kinds of effects are needed and these require at least one more track. This is for sounds like people walking around, knife and fork clatter on plates when people are eating, and one hundred and one sounds to represent other every day happenings. Sounds used for this purpose are seldom what you actually see.

In the days of dramatic radio presentation sound effects were produced by coconut shells, gravel, pieces of wood, sheets of metal, and all kinds of props. Tricks of this kind are still used for producing sound tracks for movies. Sound produced in this way always seems to be more realistic than picking up the actual sound with a microphone. For this reason it is necessary to make a complete sound track for the required effects, separately, and then dub it in, with dialogue, music, solo and other parts of the film.

In making a final composite for producing a stereophonic picture the same source tracks are used, but the final three- or four-channel tracks, or the coded arrangement, are produced by a process called "panning". This consists of mixing the different individual tracks in different proportions onto the separate final tracks. This means it is possible to make the sound of people using knives and forks come from the right side of the screen, while the sound of the orchestra or cabaret on stage can come from the left, or any other desired combination that may be required by the picture presentation.

In conclusion then, stereophonic sound for the movies has the same *basic* problems as for the home. But the best solutions, both in making the sound tracks and in presenting them, can differ considerably. All sound, intentionally or not, is stereophonic. But sound recorded with this intention *can* produce a much better illusion, provided we know what we are trying to do, and the best approach to use, according to the many things that influence it.

So stereophonic sound is quite a story. That 3-D picture analogy is quite inadequate to explain the tremendous variety of different subtle effects that sound waves can produce. Our surroundings, particularly the listening room (livingroom *or* auditorium) affect our impressions far more than we often realize. For this reason it is impossible to generalize about the best way to produce stereophonic sound.

The experiment that started in 1881 still has some way to go before we can truthfully say we know all about it. Systems so far developed represent a tremendous advance, but this form of listening pleasure in the home may still be expected to show further improvement. The advent of stereophonic sound is just one more phase in the quest that is the pursuit of high fidelity.

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