Chapter 2
Sound Waves and Electrons

SOUND is vibration, and this is true whether it be hi-fi, low-fi, or medium well done fi. The physicist would add that the vibration takes place in an elastic medium, which means that sound can travel through metal, wood or water, for example. But for hi-fi purposes we are primarily concerned with sound that travels through the air.

Sound can also be defined as the sensation produced in the brain by these vibrations. When we hear live sound, several things are happening. First, some object is vibrating and causing rapid changes in the atmospheric pressure surrounding us. The ear drums detect these vibrations, and through a rather complex process pass a message along to the brain.

In the case of hi-fi, we are adding other elements into the middle of this chain of events. As we noted in Chapter 1, the sound originally produced is converted to an electrical signal, and then converted once again into still another form for transmission or storage. Then in reproduction it is converted back to electricity, and then finally back to sound, which is transmitted through the air from the loudspeaker system to our ears.

This then is the complete hi-fi system, from the original production of sound all the way through to the auditory sensation in the brain of the listener. As a high fidelity enthusiast, you should know something about the hearing apparatus which enables you to enjoy high quality sound reproduction.

The ear is regarded as having three parts, known as the outer ear, middle ear, and the inner ear. The outer ear is the part normally visible at either side of the head, along with the canal which directs the sound inside. Stretched across the inner end of this canal is the membrane known as the ear drum, which marks the beginning of the middle ear.

When the ear drum is set in motion by sound in air, it transmits this vibration to a mechanical lever system of three bones in the middle ear known as the hammer, anvil and stirrup. At the end of the stirrup is a second and much smaller membrane, known as the oval window, which is the entrance to the inner ear.

Because of the lever action of the bones in the middle ear, and the relative sizes of the two membranes, the pressure exerted upon the inner ear is 30 to 60 times as great as that of the outer air striking the ear drum. The inner ear is a snail-shaped cavity in the skull, which is filled with a liquid. The vibrations set up in the inner

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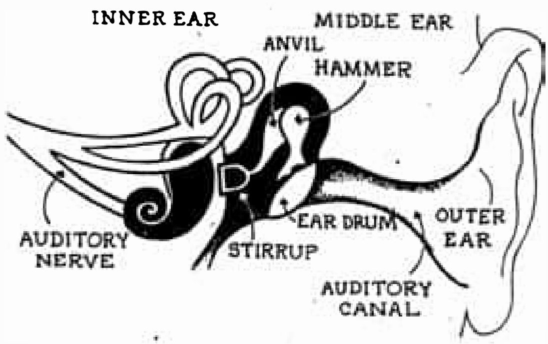
Above, Bruno Walter conducting Beethoven's 9th Symphony with full orchestra and chorus. This work is one of the most demanding; only the best equipment is recommended for its reproduction.

Right. Simplified drawing of the human ear shows the intricate design that makes hearing possible. Sound waves striking eardrum are magnified 60 times by small bone levers before reaching brain.



ear are transmitted through this liquid to the brain over a network of tiny nerve fibers.

There have been many theories concerning the precise way in which the nervous system converts sound waves in air into an auditory sensation in the brain. But nobody knows for certain even yet just how it happens. The important thing to us is that this rather wonderful thing does happen, and that it is an integral part of the overall hi-fi process.

There are many sounds in the air right now which you are unable to hear. The footsteps of a fly, for example, have been picked up and amplified by sensitive microphones and sound equipment, but no human ear has ever heard them without the aid of these artificial devices.

Sounds such as these, which are not loud enough to be heard, are said to be below the threshold of audibility. Above this threshold, the ear responds to a wide range of sound intensities, but they can ultimately become loud enough to cause pain. The point at which this happens is called the threshold of feeling.

There are essentially three classes of sound, known as noise, speech and music. The rumble of a subway train, the clatter of horses' hoofs, the clap of thunder, the crash of breaking glass: all of these are

Below, soprano Eileen Farrell and conductor Max Rudolph. The human voice is just another musical instrument and has similar overtones, vibrations.

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Fine string instruments, such as the violin Zino Francescatti plays, reach a frequency of 18,000 cps. Bass viols go down to about 40 cycles.

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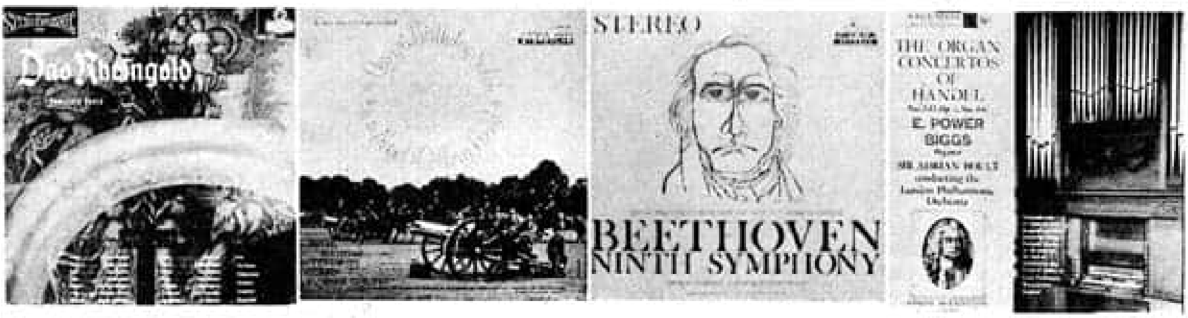


Today's modern recording techniques capture every audible note in the sound spectrum. The records above, Das Rheingold, London; Queen's Birthday Salute, Vanguard; Beethoven's Ninth, Decca; and Organ Concertos of Handel, Columbia, are especially fine examples of the record maker's art in high-fidelity sound.

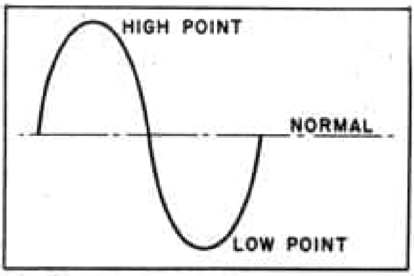
sounds which have no definite pitch. We can tell whether they are high or low, but cannot asign them a comparable note which might be struck on the piano.

Any such unpitched sound is called noise. Noises are actually mixtures of great numbers of tones, with no single one predominant enough to establish a definite pitch. Many noises therefore encompass practically the entire audible range. For this reason, various types of noises are often used to test the response of hi-fi systems.

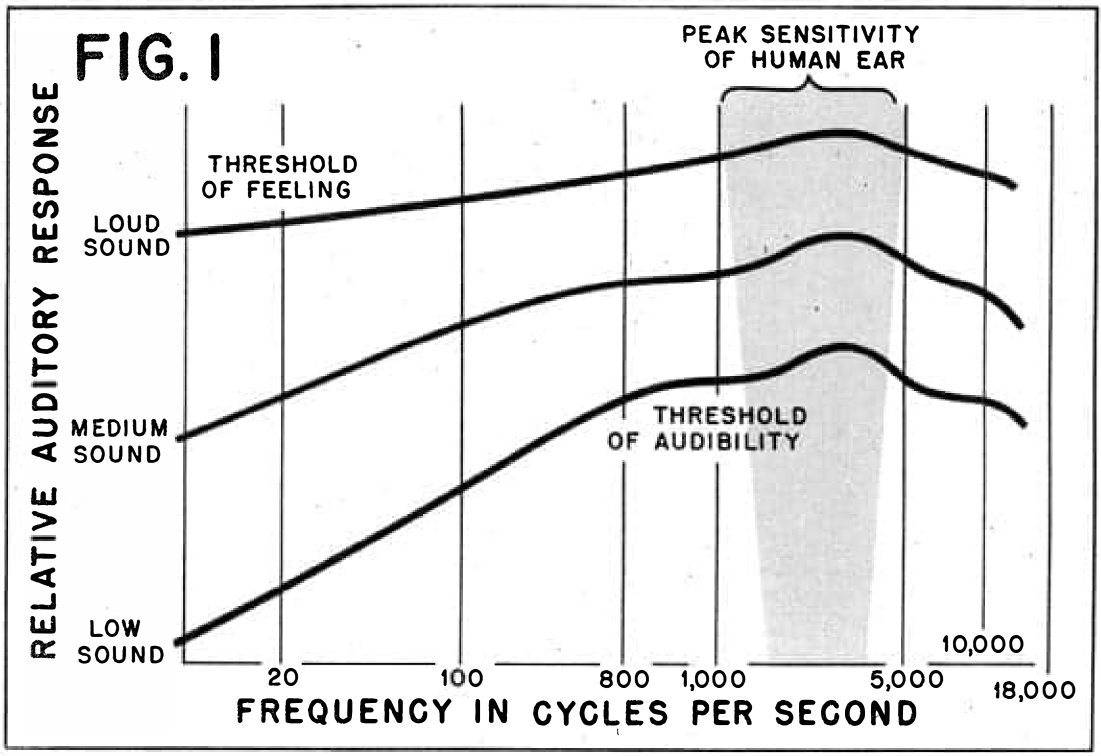
Speech sounds are those produced by the human vocal organs. The pitch of speech is more or less definite, with the average female voice being about an octave above that of the average male.

The vowel sounds, which are the most powerful in speech, are produced by employing the lungs as a bellows to produce an air stream, which in turn is set into vibration by the vocal cords. The unvoiced sounds, such as *f, k, p, ch, sh* and *th,* don't involve the vocal cords at all, but are produced by directing the air stream through small openings or over sharp edges of the teeth, lips and tongue. The voiced consonants, such as *h, w* and *y,* are really only special ways of beginning the vowel sounds, and are produced by a combination of the vowel and consonant processes.

Musical sounds are those in which the rate of vibration is uniform. Thus they have a definite pitch. Because the instruments of the orchestra have a greater overall tonal range than does human speech, and because speech is still intelligible over a low-fi telephone circuit, it is often thought that music alone provides the acid



Left. Drawing of one cycle as shown on oscilloscope. Number of cycles per second gives frequency. Below. Illustration of theory that the louder a sound the wider the range of frequencies audible to the human ear. At low volume only a small area between 2,000 and 4,000 cps is clearly discernible..At loud volume, broader band is covered.



test of hi-fi system performance. This is not true, however, for speech and noise make equally stringent demands on the system.

When we speak of pitch, we refer to the relative position of a tone on the musical scale. In hi-fi work we often use a closely related term called *frequency.* This is the rate of the vibrations of which all sound is comprised.

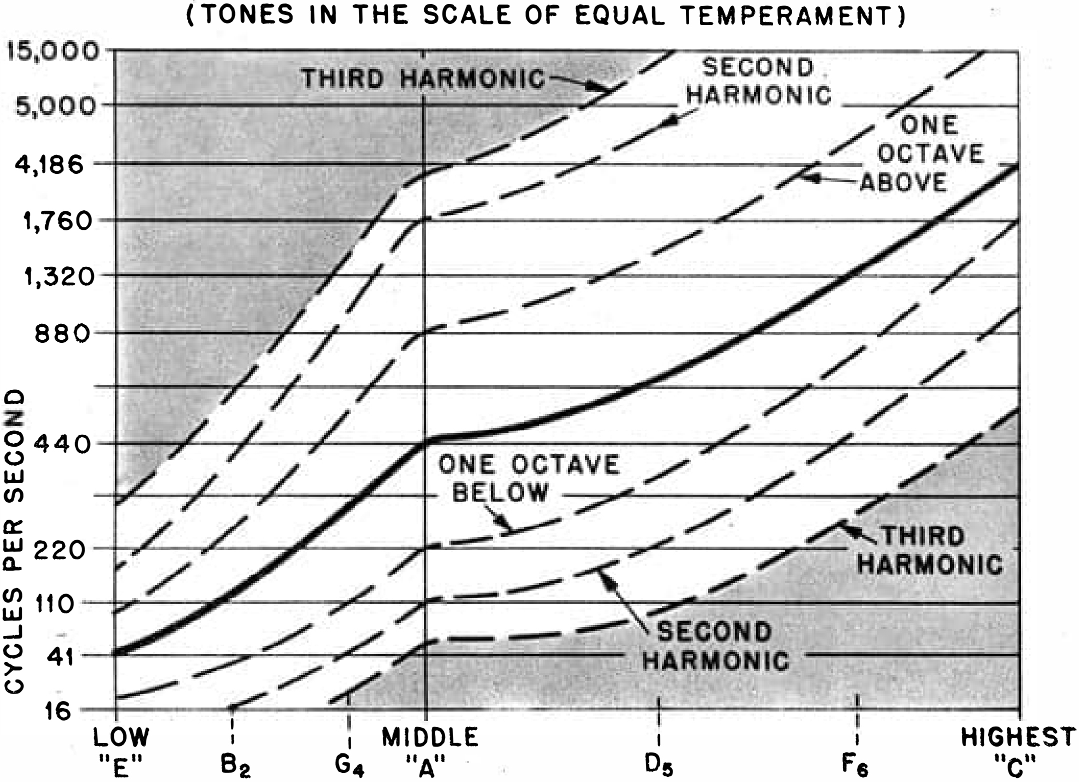
Sound vibrations in air are actually variations in air pressure, both above and below normal atmospheric pressure. Each full movement of the pressure, from normal to the highest peak and back to normal, and then from normal to the lowest peak and back again, is called a *cycle.*

The number of cycles a sound wave goes through in a given period of time is known as the *frequency.* Since one second is usually the period of time used, frequency is normally measured in terms of cycles per second, abbreviated cps. It is important to remember that, whenever you hear an audiophile speak of cycles, it's almost certain that he really means *cycles per second.*

The higher the musical pitch, then, the greater will be its number of vibrations in cycles per second. Middle *A,* for example, is 440 cps, while the low *E* string on the bass viol is pitched at about 41 cps, and the highest *C* on the piano is tuned to 4186 cps.

The differences in pitch between musical tones is known as an *interval.* While musicans often speak of intervals of thirds, fifths, ninths and such, a fairly important relationship to remember for hi-fi purposes is the *octave.*

This is the two-to-one ratio between



As shown above, although middle A, for instance, has frequency of 440 cps, the harmonics go three times as high and three times as low. See text for fuller explanation how this affects hi-fi equipment.

tones. Thus with middle *A* pitched at 440 cps, going up the scale to just twice the frequency, 880 cps, we find another *A*, this one said to be an octave above the first. Similarly, going down the scale to 220, we find another *A* below middle *A*. Each note on the scale, in fact, is separated from those of the same name adjacent to it by exactly twice the frequency, or one octave.

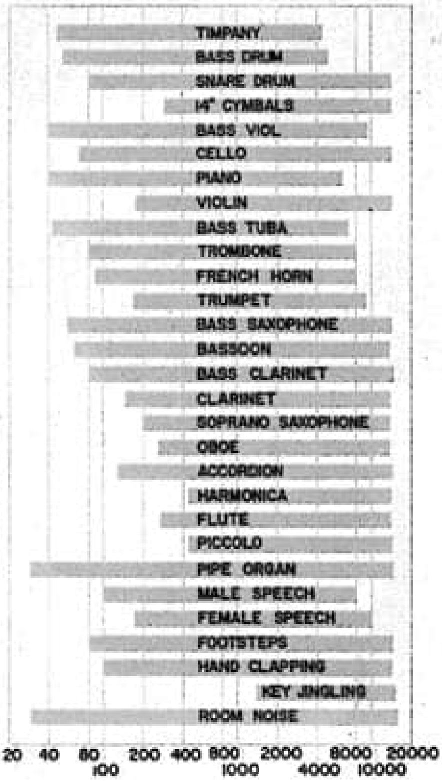
The lowest tone produced by any musical instrument is about 16 cps, while the highest is in the neighborhood of 5,000 cps. This is a span of about eight octaves. It would seem at first glance that a system capable of reproducing all of this eight-octave range should be quite adequate for sound reproduction. But the fact is that such a system would not be very high fidelity at all, as we shall see.

We all know that there is considerable difference in the sounds produced by the various instruments in an orchestra, even if all of them are playing the same note. If this were not so, there would be no point in having all those different voices.

But if *G* below middle *C*, for example, is always 196 cps, then there must be still another factor determining the differences in the sound *timbre* or *tone quality*. The reason for this is that no instrument produces an absolutely pure tone. That is, when *G* is struck on the piano, there are several frequencies produced in addition to 196 cps, the *fundamental* frequency.

These extra vibrations are known as *overtones* or *harmonics*, and their frequencies are normally multiples of the fundamental. In other words, the important harmonics are two, three and more times the fundamental.

Now the main difference in the timbres of the instruments of the orchestra lies in the harmonics these instruments produce, and the relative strengths of each as compared to the fundamental. These characteristics will also often vary with even the same instrument. When the *G* string of the violin is bowed, for example, all of its harmonics up to the thirteenth are stronger than the 196-cps fundamental. The *E* string, on the other hand, has third, fifth and eighth harmonics about equal to



Frequency range of various instruments, human speech, etc. At right is drawing of piano scale.

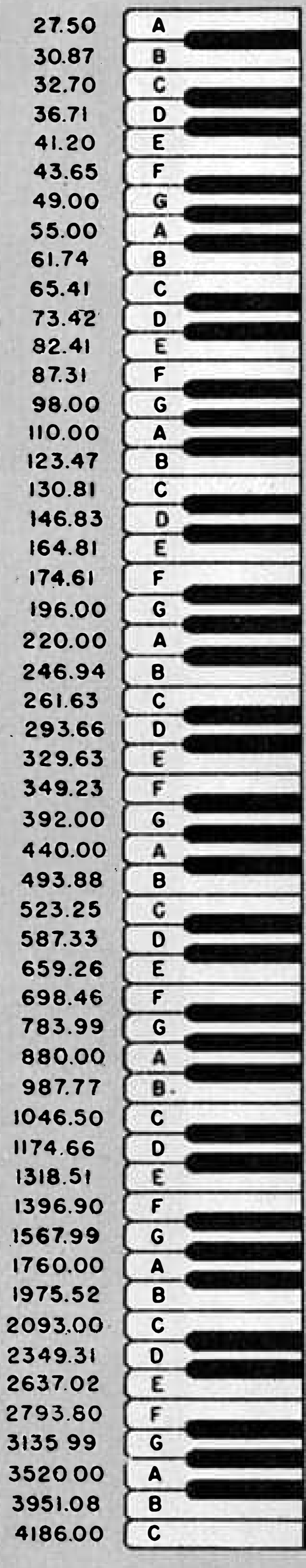
the fundamental, while the others are less.

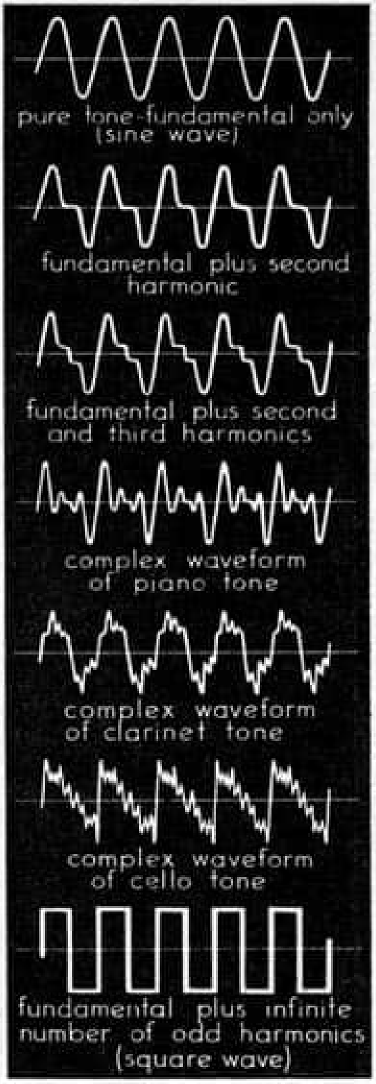
These facts are very important to us in hi-fi listening, for it is by these harmonics that we distinguish the characteristics of the various instruments. It has been shown experimentally, in fact, that the human ear must be able to hear at least up to the third harmonic of a tone to be able to identify the instrument producing it.

If we hear an instrument playing a middle *A* at 440 cps, for example, we will be unable to tell what instrument it is unless we can hear all the way to the third harmonic, or 1,320 cps. And this fact gives us a major clue to one of the important benchmarks of hi-fi system performance.

We have already noted that the highest fundamental frequency of any present musical instrument is about 5,000 cps. Then in order to hear the third harmonic of this tone, we require a system which is able to pass three times that figure, or up to around 15,000 cps.

So there we have it. A good hi-fi system must be able to pass the lowest musical note, about 16 cps, and the third har-





These vertical patterns are made by sound on the screen of an oscilloscope. Harmonic components of a tone give the quality of the individual instrument.

Modern jazz, trombones, trumpets, drums, etc., demands a full-range system for true fidelity.

monic of the highest musical note, or around 15,000 cps. And it also must be able to pass everything in between as well. The actual frequency range encompassed by the hi-fi system is known as its *frequency response.*

When the frequency response of a hi-fi system is spoken of as *flat,* this means that it does not alter the relative loudness of any of the tones within its range. If it does so, it, of course, changes the delicate relationship between harmonics and fundamentals of the instruments, and thus distorts their quality.

The loudness of a sound is its relatively high intensity as perceived by the human ear. Loudness is expressed in *decibels,* and is essentially a ratio of two acoustic powers. Thus we might say that the maximum loudness of a bass saxophone is about 8 decibels (abbreviated db) above that of a clarinet. Similarly, we might say

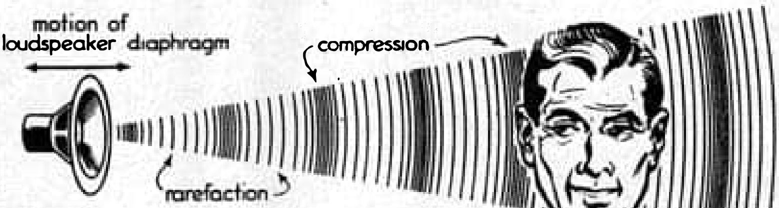
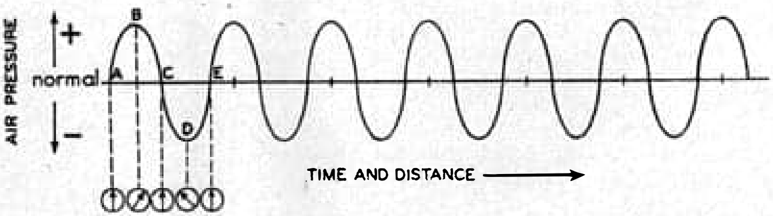


FIG. 2



Sound is caused by changes in air pressure. One ear in drawing is shown in compression area, and one ear in the air's rarefaction region.

that turning up the volume control on our hi-fi system has increased the output by, say, 5 db.

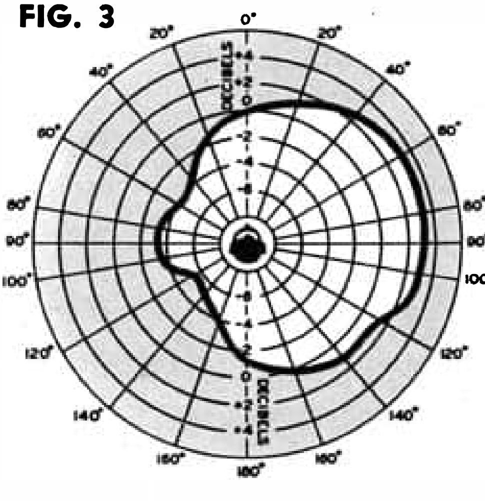
There is good reason for expressing loudness as a ratio rather than an absolute value. Suppose, for example, that by turning up the gain on a given amplifier we increase the output from 2 to 5 watts. This 3-watt increase is in a ratio of 2.5 to 1, or 4 db.

But if another amplifier is increased from 10 to 13 watts, the same 3-watt increase is in a ratio of 1.3 to 1, or barely over 1 db. The 4-db increase would be quite significant to the ear, while 1 db would probably be entirely unnoticed. It is therefore essential that we know our starting point before we can analyze intelligently any change in sound level.

The frequency range from 16 to 15,000 cps, which we established as minimum for hi-fi requirements, is also just about the range of the average human ear. But oddly enough, the ear is nowhere nearly flat in its response. It is markedly more sensitive in the region around 3,500 cps, and much less so in the extreme bass and treble regions.

This graph shows how each ear responds if a sound of constant intensity is moved from front to back around the right side of listener's head.

FIG. 3



Furthermore, this response varies with the loudness of the sound. When the sound is extremely loud, for example, the midrange peak and the droops at either end are much less pronounced. This is shown in Fig. 1. Since the ear itself distorts in this fashion, it is sometimes difficult to understand the necessity for a perfectly "flat" hi-fi system.

The reason is that the system must do nothing to alter the relationship between the original sound and its effect on the ear. The fact that human hearing is not "flat" is of no serious consequence, so long as the sound reaching the ear is identical to that originally produced.

We don't actually have such perfection in hi-fi yet, as we observed in the previous chapter, but at least we have identified the target. From our knowledge of human hearing, we can conclude that the perfect hi-fi system would exhibit the four following basic characteristics:

1. The full frequency range of the human ear
2. The full loudness (dynamic) range of a symphony orchestra
3. Absolute freedom from distortion of any sort
4. Absolute freedom from extraneous noise

Point 1 is quite adequately fulfilled by today's systems. For the most part, this is also true of point 2. Point 3 still has a few percentage points to go yet before distortion reaches zero, while the noise condition, point 4, is reasonably good.

Photo below shows members of the Budapest String Quartet recording with guest artist Walter Trampler. Inset, Alexander Schneider, Boris Kroyt, and Joseph Roisman, listen to a playback of the tape. Performers are often among the most critical listeners.

Columbia Records

The Stereo Story

Up to here in this chapter we have been referring to the human "ear," in the singular. Although this might be taken in the literary sense to mean the human hearing system collectively, in this case it would be better to take the term literally.

For the human hearing system actually comprises two ears, and the combination gives us still another characteristic known as *directivity*. Because we have two ears, we can tell the direction from which a sound is coming. This can be understood rather readily by reference to Fig. 2.

Now let's see what happens to you when you're within earshot of that speaker. The varying air pressures set up by the vibration of the speaker cone will push and pull against each of your ear drums. These vibrations in turn set in motion the complicated process of hearing which ends with your brain receiving the sensation of sound. Since your ear drums are separated by the width of your head, the sound must travel different paths between the source and each ear.

In Fig. 2 you see that your right ear is closer to the speaker. Thus the sound will get to it before it reaches the left ear. This is so because your head is turned at an angle relative to the speaker.

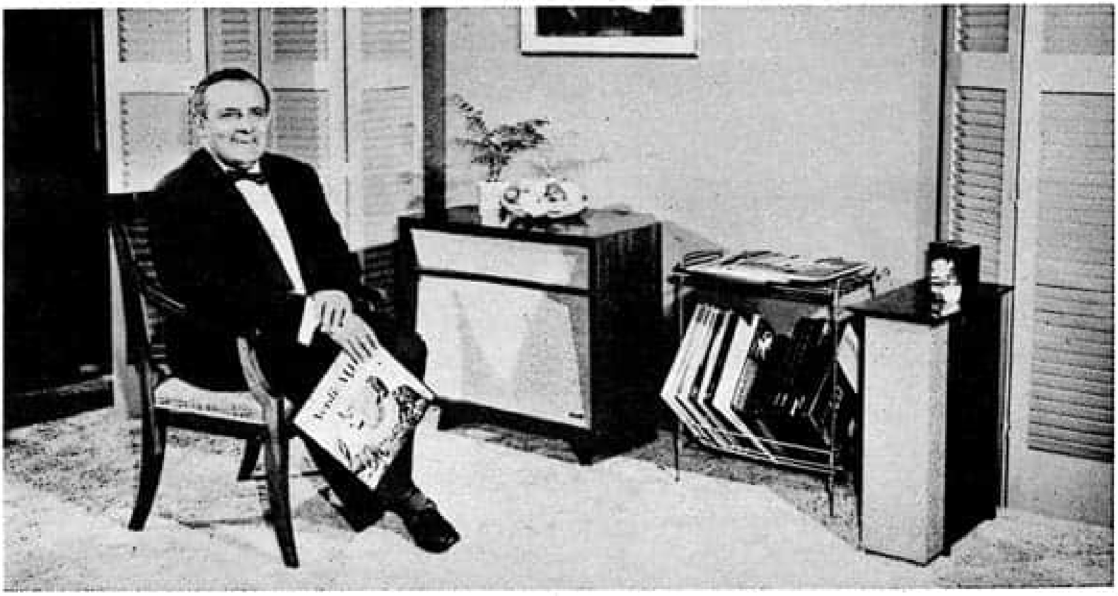
Now remembering that distance traveled equals (rate x time), let's see how that applies to the sound from the single speaker striking your ears. The rate—in this case the velocity of sound—will be constant. The distances will be different. Therefore the *times* will not be the same. In fact, we can generalize and say that, except for those circumstances when the sound is directly in front of or directly behind you, the arrival times of the sounds at the ears will *never* be the same.

This time differential is in turn responsible for a difference in *phase* of the sounds reaching each ear. Notice in the drawing that while the right ear is in a high pressure area, the left ear is in a low pressure trough. Since sound is a wave motion

Sound Waves and Electrons

Baritone Leonard Warren is another exacting listener, one of the many artists who have hi-fi equipment. Stereo is his choice because it best captures the full range of voice and orchestra.

University Loudspeakers, Inc.



moving constantly outward from the source, in another instant the compression part of that wave will have moved on to your left ear, while your right ear will be in a rarefied area.

These differences in time and phase between sounds striking each ear will give your hearing system the clues it needs to estimate both the distance and direction of the sound source. You can get some idea of the ability of your ears to sense direction by referring to Fig. 3. This shows how each of your ears would respond if a sound of constant intensity were moved from front to back around the *right* side of your head.

The ear on the same side as the source—in this case the right—of course hears more. But the loudness of the sound does seem to vary as the sound moves around. The response of the ear is almost heart-shaped, with greatest sensitivity to sounds arriving from a little ahead of abeam, or about 75 degrees.

The opposite ear, naturally, hears most when the sound is closest to it, which would be directly forward or behind.

These differences in directional responses of the ears are evidently due to the physical construction of the external parts of the ear and of the head. When combined with the differences in time and phase of sounds arriving at each ear, there is sufficient information for the brain to discern distance and direction.

This is strictly true, however, only at the higher frequencies. As we have already noted, the differences in time and phase becomes scarcely noticeable at the longer wavelengths around 300 cps and lower. Just what this fact means in terms of a hi-fi stereo system is still a matter of some controversy. One school of thought argues that two wide-range channels are not necessary, that the bass can all come from one loudspeaker since it is basically non-directional in character. Others say that it's a nice theory which doesn't prove out in practice. We'll examine both sides of this argument in Chapter 9. •