

THE EFFECT OF HEAD DIFFRACTION ON SOUND RECORDING

Theodo

by

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ABSTRACT

The concept of the binaural loudness sensation 2. person experiences in ^a sound field is discussed. An assumption is made which relates ^a single variable to this binaural sensation, for purposes of making tests and comparisons. The goal is to find ^a recording method which, for pure tones, records on ^a single channel a signal which has the same power density spectrum as that experienced by ^a listener, according to the assumption.

^A new recording technique is submitted for purposes of evalu8tion and comparison. The remainder of the thesis consists of the data from these tests and ^a discussion of the results.

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CHAPTER I

INTRODUCTION

1.1 Statement of the Problem

The problem of recording music or other types of sound is neither new nor is it easy. There has not yet been a sound recording technique developed which is totally satisfactory to everyone. Probably this statement can still be made for ^a number of years in the future. The important point is this: there are many techniques by which sound may be recorded. This thesis concerns a comparison of two of them.

^A rather simple and common approach to this problem is the placement of a single microphone in the presence of the sound field to be recorded. A tape recorder or similar device is used to permanently record the character of the sound. Thus, an electrical signal may be obtained which resembles the sound pressure on the surface of the microphone. Further discussion will assume a perfect microphone, i.e., one which gives an output voltage exactly proportional to the sound pressure in the sound field being recorded, where this pressure is the pressure

that would be there in the absence of the microphone. In terms of power density spectrums, the recorded spectrum would be identical to the spectrum of the nressure which would have been there in the absence of the microphone. This recording technique will be referred to as the single free microphone technioue.

In reality, there are at least two types of microphones. One type has a frequency response which reads flat when tested in a sound field whose pressure is actually flat without the microphone being there. For this test the sound field is normally incident on the surface of the microphone. This type of microphone is said to have ^a flat free-field calibration.

The other type of microphone has a frequency response which reads flat when the pressure at the surface of the microphone is actually flat. The pressure response of this microphone is said to be flat. The difference between these two types of microphones is in the amount the microvhone distorts the field. The first type compensates for its distortion. The second one does not compensate for its distortion.

Ordinarily, the purpose of recording sound is to be able to reproduce the sound field so that a person or persons might get the same acoustic

sensation during the reproduction that he would have gotten had he been present when the recording was made. This poses the question: what is the nature of the sensation a person gets at a concert, and does it differ from that of the microphone. if some method existed for comparing these two sensations?

An acoustic phenomena known as "Binaural Summation of Loudness" is mentioned by Licklider (5). This phenomena seems to indicate that the binaural loudness sensation ^a person gets in the presence of pure tones is a measure of the sum of the intensities at the two ears. If a given intensity were present at both ears, an intensity present at only one ear would have to be twice that great in order to receive the same loudness sensation for both experiments.

This loudness sensation under discussion is a nebulous thing. It is ^a measure of the total effect ^a sound field has on ^a person, considering both ears. Yet it is ^a singular thing, so it must be treated accordingly. That is why it is necessary to assign ^a single intensity to represent it.

The assumption is now made that, for pure tones, the binaural loudness sensation is ^a measure of the sum of the intensities or energies sensed by the two ears. Further discussion will depend on this assumption,

^a logical question now is, "What affects the intensity sensed by the ears?" There are two areas which affect the hearing process: first, the nature of diffraction which sound waves experience while reaching the entrance to the auditory canal; and second, everything else from that point on through the canal and through the remainder of the sensory mechanism.

Wiener (7), in his studies on the diffraction of sound waves by the human head, observed a phenomena which is pertinent to this problem. The pressure distribution in the auditory canal is largely independent of the orientation of the listener with respect to the sound source. Once the sound waves reach the entrance to the auditory canal, the geometric relation between the position of the head with respect to the sound field is no longer ^a consideration. Further dependence concerns only frequency.

The significant quantities, then, are the pressures at the entrances of the ear canals, and the diffraction patterns of the head. This thesis will be concerned with these quantities.

The goal now is to record some signal which for pure tones has the same spectrum as that experienced by ^a listener, according to the assumption. The sound must be tones, because the assumption concerns

intensity which is nonlinearly related to the pressure and does not work for random waves.

The intensity of sound at the entrance to the left ear will be defined as I_1 , the right ear, I_2 , and the total intensity according to the assumption, I_7 . The pressure of the sound at the left ear will accordingly be defined as P_1 , the right ear, P_2 , and a pressure associated with I_3 , P_3

According to the above assumption:

$$
I_3 = I_1 + I_2
$$
 Eq. 1.1

for pure tones.

From simple acoustic theory:

$$
I = kP^2
$$
 Eq. 1.2

where k is a constant.

Then substituting Eq. 1.2 into Eq. 1.1 for the three intensities:

$$
kP_3^2 = kP_1^2 + kP_2^2
$$
 Eq. 1.3a

$$
P_3 = KP_1 + KP_2
$$

\n
$$
P_3^2 = P_1^2 + P_2^2
$$

\n
$$
Eq. 1.3b
$$

 $Eq. 1.3c$

from which $P_3 = (P_1^2 + P_2^2)^{1/2}$

Since the sounds are pure tones, the pressures
$$
P_1
$$
, P_2 , and P_3 , may represent either the rms value or the peak value of the pressure. The reason for this statement will be given when the test equipment is discussed.

10.

or

It will be noticed that this assumption neglects any sensation due to phase, and concerns only magnitudes, Admittedly, this is an assumption, but it is thought to be valid. At any rate, this thesis will be concerned only with magnitudes and not phases. If it turns out that phase is an important consideration, it will be studied in subsequent investigations by others.

A brief summary of what has been said so far is probably now in order. It was assumed that the binaural loudness sensation ^a person gets in ^a sound field is ^a measure of the sum of the intensities sensed by the two ears. It was concluded that the amplitudes of the pressures at the entrances to the auditory canals contained the significant information. Specifically, for pure tones, the sum of the energy at the two ears may be used as a measure of the binaural loudness sensation for testing purposes.

One possible method of combining the signals present at the two ears is simple addition.

$$
P_4 = P_1 + P_2 \qquad \qquad Eq. 1.4
$$

where P_4 is this sum. A microphone can be placed at the entrance to each ear canal in ^a dummy head, these two signals added, and the result recorded. At this point it certainly is not apparent that this is ^a satisfactory solution. However, diffraction

studies can be made on the dummy head, using tones, thus allowing a comparison between P_3 and P_4 . Also, comparisons can be made between these pressures and the pressure detected by a single free microphone. These tests and comparisons will constitute the remainder of the thesis.

1.2 Background for the Problem

The nature of this problem is such that it has little background. As far as is known, no one has done experimental work on the diffraction of sound waves around a dummy head, or its effect on sound recording utilizing a dummy head. However, work has been done on the diffraction of sound around human heads. This is pertinent, since it is permissable to see just how closely the dummy head resembles a human head, at least acoustically.

Fletcher and Munson (4) studied the sensitivity of the human ear to different frequencies. They observed that the ear is not equally sensitive to all frequencies. Equal intensity sound fields appear louder in the mid-frequency band that in the low or high freguency bands. Below a certain low freguency and above ^a certain high frequency. sounds cannot be heard at all, no matter how great the intensity. This effect alone insures that the spectrum of a sound recorded by a single free microphone will differ from that experienced by a listener in the same location.

Sivian and White (6) in 1932 studied minimum audible sound fields in general. In particular,

they studied minimum audible sound fields as ^a function of the angle of incidence of plane waves. These results are given as a series of polar plots. indicating the relative sensitivity of monaural hearing, one polar plot for each of ^a number of representative frequencies. Although this data concerns minimum audible sound fields, it should be ^a measure of the directivity of monaural hearing at any listening level.

Sivian and White also calculated the pressure which would be produced on the surface of ^a rigid sphere by an incident plane wave as a function of the latitude,

In their discussion, Sivian and White give ^a formula which may be used to calculate the binaural minimum audible field when the observer is exposed to ^a diffuse sound field in which all wave fronts are vertical, all azimuths equally probably, all amplitudes equal, and all time phase angles distributed at random. For such ^a field the resultant binaural minimum audible field is equal to the binaural minimum audible field observed when the listener is facing the source of sound F_0 multiplied bv the factor R. As given, the formula for ^R contained ^a printer's error. The correct formula

for ^R is now given as:

$$
R = \left[(f_0^{2\pi} F_\theta^2 \ d\theta) / 2\pi F_0^2 \right]^{1/2} \qquad Eq. 1.5
$$

where F_{θ} is the magnitude of the minimum audible field as ^a function of angle.

Wiener (7) , in 1946, studied the diffraction of a progressive sound wave by the human head. His results are summarized in a number of frequency
response curves, one curve for each 45⁰, for the human head at discrete angles with the sound waves. These curves give the ratio of the sound pressure at the entrance to the auditory canal to the pressure existing in the region before the person entered. The curves only cover the frequency range from 200 cps to 6,000 cps.

CHAPTER II

METHOD OF EXPERIMENTATION

2... Description of Equipment

^A somewhat extensive description of the equipment is necessary because it differs from that used for most of the past experiments in this field.

This is called narrow band noise. It consists of I, a slightly different form of sound will be used. Although pure tones were discussed in Chapter white noise which has been filtered so that only a narrow segment of frequencies centered about some center frequency, is present.

This narrow band noise techniaue has one major advantage over pure tones. It eliminates standing waves which usually are troublesome. Also, for experiments ultimately concerning random waves like music and the like, it is thought that it is ^a better indication of the response of human ears.

The narrow band noise is sufficiently like pure tones so that it still may be used in the experiments involving the sum of energies without appreciable error.

Experimental data was taken with the General Radio Type 1521-A Graphic Level Recorder, hereafter called the graphic recorder. This device produces ^a permanent ink record of the level of ac voltages. at frequencies throughout the audio band. The graphic recorder responds to the rms level of the input waveform. It may be used in conjunction with an audio oscillator or with the continuously variable narrow band filter which will be discussed later. The graphic recorder is geared with the oscillator or filter so that the oscillator frequency or filter center frequency is continuously swept across the audio band while the graphic recorder registers the desired response. The graphic recorder is particularly useful because it allows large amounts of data to be taken swiftly and accurately.

Narrow band noise was obtained by using a white noise generator in conjunction with the General Radio Tyve 1554-A Sound and Vibration Analyzer, hereafter called the sound analyzer. This device is ^a continuously variable narrow band filter. The filter response characteristic is shown in Fig. 2.1.

 17_o

Fig. 2.1 Filter Response Characteristic of General Radio Type 1554-A Sound and Vibration Analyzer

The response of the sound analyzer is down 3 db at $+$ 4% of the selected frequency. At $1/2$ and 2 times the selected frequency, the response is down over 40 db.

Since the sound analyzer bandwidth is proportional to the center frequency, white noise appears to have a slope of +3 db per octave, on the graphic recorder. However, compensation is simple. ^A pink filter is connected to the output of the white noise generator. This filter attenuates the white noise ³ db per octave over the audio band, to yield pink noise. Pink noise then has zero slope on the graphic recorder.

Two $1/2$ inch condenser microphones Type 4133 nade by Bruel & Kjoer were used. This is ^a high quality microphone designed for precise sound measuring purposes over a wide frequency range. It has a flat frequency response for sound waves impinging perpendicularly on the diaphragm. The specified free-field-response for that condition $is + 2 db from 20 cps to 40 kc. Actually it is$ well within those specifications. The calibration curves for this microphone are shown in Fig. 2.2. The two microphones are identical within $1/4$ db everywhere.

Two loudspeakers were chosen because of their relatively flat frequency characteristics. One speaker was used for frequencies between 100 cps and ¹ kc. Another speaker was used for frequencies between 1 kc and 20 kc.

2.2 Procedure

All sound measurements were made in the M.I.T. Anechoic Chamber. This permitted a good plane wave to be generated, with little reverberation. The dummy head was mounted on a two foot rod which could be rotated by remote control. The speakers were at ^a distance of ten feet from the head. The low frequency speaker was directly behind the high frequency speaker, so that they both lay in the same horizontal plane with the dummy head, Polar plots were made with each speaker at a frequency of 1 kc and these curves were within $1/2$ db everywhere. indicating the consistent data due to speaker placement.

Single free microphone curves were taken on the speakers with the microphone in the extreme positions it would occupy in rotation of the head. No significant variations occurred, indicating ^a uniform sound field in the vicinity of the head.

In early experiments, Altec-Lansing 21-D microphones were used. These microphones gave trouble because of heating, since they were imbedded in the dummy head for long periods. Their frequency response changed excessively with temperature. However, with the 4133 microphones, heating was

no problem because of their low temperature coefficient.

Two different dummy heads were studied before the one used for this data was selected. The final selection was based on which one gave results most similar to those of Sivian and White (6) . It should be mentioned that extensive tests were also made on the other dummy head which bore a much closer physical resemblance to human heads. Although it gave results generally like those of Sivian and White, it was not as good an acoustic model as the one used. The acoustically inferior head did not shade the high frequency sounds as well as the other. This was thought to be due to the difference in material covering the dummy heads. The inferior head had a thick layer of soft rubber covering ^a plaster of paris base. The head which was used had ^a very thin covering of soft rubber over ^a balsa wood base. Evidently, the thick rubber layer conducted sound waves excessively.

Basically, there were three different circuit connections as shown in Fig. 2.3. For curves labelled "Souare Root of Sum of Squares of Two Ears", the square law device accepted as inputs the output of the microphones placed in each ear. The square law device actually gives an output proportional

Fig. 2.3A Block Diagram For "Square Root of Sum of Squares of Two Ears" Experiments

Fig. 2.3B Block Diagram For "Sum of Two Ears'' Experiments

Fig. 2.3C Block Diagram For "Right Ear Only" Experiments

to the sum of the squares of the two inputs, so an 80 db potentiometer was used in the graphic recorder to take the desired square root. This allowed ^a direct comparison between these curves and those not involving the square law device, where a 40 db potentiometer was used.

The squre law device does not square the incoming signal directly. It first peak-rectifies the incoming signal and then squares it. For sine waves, this yields an output proportional to the sum of the squares of the amplitudes of the input signals. This is why it was mentioned that P_1 , P_2 , and P_3 could represent the peak values, or amplitudes, of the pressures.

For curves labelled "Sum of Two Ears", this sum was obtained by connecting two transformers in series. Each microphone had a transformer output, so it was a simple matter to connect these in series, and amplify the result.

The final frequency response curve of the dummy head at each angle was obtained by subtracting the loudspeaker curve of the single free microphone from the loudspeaker curve of the dummy head at that angle. This gives the ratio of the response of the head at that angle to the single free microphone response.

The frequency response curves of the dummy head to random incidence waves were calculated by means of Eq. 1.4. Each polar plot was first squared, and then redrawn to linear rather than logarithmic scale. These curves were then integrated by means of a planimeter. Separate calculations were made at ¹ kc for each speaker as ^a check. The final result differed by less than 1/2 db for all three curves.

For all graphs consisting of the three curves representing the three methods of recording, the three curves were all arbitrarily set at ⁰ db at 100 cps for purposes of comparison.

CHAPTER III

EXPERIMENTAL RESULTS

3.1 Frequency Response Curves and Polar Plots

The following pages contain the results of the experimental investigation of the dummy head. First are the polar plots. Next come the frequency response curves of the head at discrete angles with fhe plane waves. Finally is the curve which gives the response of the head to random incidence sound. This curve was calculated from the polar plots by using Eq. 1.5. Section 3.2 contains the discussion of these curves.

0 db is response at
100 cps

1800

 270°

360^o

at

3 kc

O db is response at 100 cps

 ϵ

 -5 -10

 -150

90°

O db is response at 100 cps

 100 cps

 -15 0^o 90^o 180° 270° 3600

O db is response at 100 cps

Fig. 3.17 Frequency Response Curves of Dummy Head at 45[°]

Fig. 3.18 Frequency Response Curves of Dummy Head at 90°

Fig. 3.19 Frequency Response Curves of Dummy Head at 135[°]

Fig. 3.20 Frequency Response Curves of Dummy Head at 180°

 \boxtimes 38.

Fig. 3.21 Frequency Response Curves of Dummy Head at 225[°]

"Right Ear Only" Legend:

Fig. 3.22 Frequency Response Curve of Dummy Head at 270°

Fig. 3.23 Frequency Response Curve of Dummy Head at 315[°]

 $42.$

Fig. 3.24 Frequency Response Curves of Dummy Head to Random Incidence

"Right Ear Only"

3.2 Discussion and Interpretation of Data

Two types of experiments were performed on the dummy head using four different recording methods. The two types of experiments are the studies on the polar response of the dummy head and the studies on the frequency response of the dummy head at discrete angles.

The first type of recording method is the single free microphone discussed in 1.1. The next recording method is the one which is thought to record the same spectrum as humans sense, when pure tones constitute the sound field. This is the "Square Root of Sum of Squares of Two Ears" technique.

The third type of recording method is the "Sum of Two Ears" technioue. This method was proposed as a subject for experimental investigation.

The fourth recording method is the "Right Ear Only" technique. This method was studied as a check on the other methods, and so a comparison could be made with monaural studies made on humans to determine how similar the dummy head was to human heads. The signal from the microphone in the right ear was recorded for this techniaue.

The results for the last three methods are presented relative to the results of the single free microphone. The single free microphone is represented by the ^O db line on all the curves.

An explanation of the two types of microphones was given in l.1. It is apparent that the flat free-field calibration microphone should be used for the single free microphone technique, because it gives ^a reading of what would exist if it did not disturb the field,

It is not so apparent which calibration should be used when the microphone is placed in the dummy head. What is desired is a signal proportional to the pressure existing at the entrance to the auditory canal without the microphone being there. The free-field calibration is not exactly proper, because it is compensated to read the pressure that would exist in free space, and not in the ear of a dummy, since the dummy does not have the same acoustic impedance as air.

On the other hand, the pressure calibration is not proper either. It would be proper if the acoustic impedances of the microphone and the dummy head were equal.

The calibration which should be used when the microphone is in the dummy head actually lies somewhere between the two curves of Fig. 2.2.

The data in this thesis was calculated using the same calibration for all cases, namely the flat free-field calibration of Fig. 2.2. The frequency response curves beginning with Fig. 3.16 are too high by a maximum of $3 \frac{1}{2}$ db at 10 kc, and less than that at lower frequencies. The error due to this consideration is never greater than the difference between the two curves of Fig. 2.2.

The discussion will begin with the polar plots and then move on to the frequency response curves. At 100 cps, there is seen to be no change in response due to the angle of incidence. This is expected because at 100 cps the wavelength of sound is about ¹⁰ feet and the dimensions of the head are not sufficient to disturb the field. Consequently, all three curves read ⁰ db for all angles.

In the 400 cps and 700 cps curves the "Sum of Two Ears" curves are observed to dip quite ^a bit when either ear faces the speaker. At these frequencies, the sounds at the two ears tend to be out of phase, so there is a cancellation. The "Square Root of Sum of Squares of Two Ears" curves do not show this effect. That is because phase difference does not affect this technique. The "Right Ear Only" curves show the expected increase in pressure due to pressure doubling when the open ear faces the speaker, as well as the decrease in pressure due to shading when the open ear is away from the speaker.

At 1 kc the square law curve is still relatively flat, while the additive head shows marked deviations due to angle. The right ear curve begins to show the "Bright Spot" effect. When the open ear is directly away from the speaker, sound waves tend to arrive around the head from all angles in phase and add up. This phenomena can be seen to occur in the mathematical model studied by Sivian and White and mentioned in 1.2. This accounts for the buildup at 90° and 270° in the additive head curve.

Most of these same effects occur in varying degrees throughout the rest of the polar plots. The extreme variations in response which begin to show up near 0^0 and 180^0 in the additive head curves result from the fact that the signals from each ear are approximately equal in magnitude and the wavelength is short. At ^a certain angle they will be in phase and add up. Just a few degrees away they will be out of phase and almost cancel. At 90[°] and 270[°] almost all the signal comes from one ear, so cancellations do not affect the smoothness of the curve. The souare law curves are seen to remain smooth because it does not have phase cancellation.

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At 6 kc and 7 kc the right ear curves show an interesting phenomena. At these frequencies, the head hears sounds better behind it than in front of it, Sivian and White observed this in humans at 7600 cps. Also the square law curve shows that the binaural head hears better from behind than in front.

The higher frequency curves show more of the same type results. The main significance is that shading becomes extreme at these frequencies.

At 15 kc, the square law head hears almost 20 db less behind it than in front of it. That is due to the ear lobes shading almost all of the sound,

The right ear curve at 15 kc is seen to be flat in the vicinity of 90° . This is thought to be due to the noise level during the experimental NOTK.

Comparisons between the "Right Ear Only" curves and the curves Sivian and White obtained from people indicate that the polar response of the dummy head used was very similar to human heads. Of course it was not exact, but the curves had the same general form, and in most cases the magnitudes of the peaks and dips were close. Dummy heads could be built which more closely resembled humans, but this head was good enough for ^a first study.

The frequency response curves beginning with Fig. 3.16 show mostly the same effects from ^a different viewpoint. At 0° , or with the listener facing the sound source, there is no significant hree recording methods. This is to be expected, since the signals at the ears are identical. The significant point is that all three recording techniques do not record the same spectrum as a microphone. The frequencies around 3 kc are boosted over 5 db, and the frequencies near 7 kc are attenuated over 5 db.

As the head is set at different angles, the three curves differ considerably. The additive head shows the familiar dip around ⁷⁰⁰ cps explained earlier. The right ear curves fall off greatly at high frequencies due to shading. The square law curves are seen to be the smoothest , as they were in the polar plots.

It will be noticed that, beginning with 225[°]. only the right ear frequency response curves are shown. This is due to symmetry. For example, at 225° , the other two curves are the same as the ones at 135°.

As mentioned earlier, the random response curves of Fig. 3.24 were calculated from the polar plots. The right ear curve is very similar to the square

law curve, and always above it. This happens because they were both made to pass through 0 db.at 100 cps. The additive head curve is very similar to the square law curve, except that it is about 3 or 4 db lower from 600 cps out past 10 kc. This attenuation is due to the cancellation caused by the addition of a number of variables. More times than not, they will be out of phase.

The amazing fact shown by the random response curves is that the response of the dummy head is so flat. There is ^a ² or ³ db boost throughout the mid-frequency range, but this is very small compared to the variations at individual angles.

On the strength of these curves, it is possible to draw certain conclusions. If ^a recording is being made in a very directional field, corresponding to the 0° case, either the "Sum of Two Ear" technique or the "Right Ear Only" technigue gives results which are almost identical with the "Square Root of Sum of Sauares of Two Ears" results, which. were desired.

If the recording is being made in an entirely diffuse field, then the "Right Ear Only" is the best recording technique. However, simple compensation on the "Sum of Two Ears" would bring its response up to that of the "Square Root of Sum of Souares of Two Ears", and it would be ^a good method also,

The most interesting and difficult problems are in the area where the sound to be recorded is neither entirely directional or diffuse. For sound arriving at any one angle it is possible to design a filter which will make the frequency response of the additive head the same as the sum of energies head. However, this compensation will not, in general, be appropriate for other angles. It is impossible to compensate perfectly for all angles. Since the problem is to record the same spectrum for tones as the "Square Root of Sum of Squares of Two Ears" technique does, then it is impossible to compensate the "Sum of Two Ears" technique to accomplish this perfectly for anything other than plane waves. Obviously, this means that some imperfect but acceptable technique must be devised for other special cases. Factors to be considered in choosing the recording techniques for these special cases will probably be subjective ones. For example, it is possible that the "Sum of Two Fars" technioue might incorporate ^a filter for compensation which resulted in an error which was not distinguishable to the ear. If this were not possible, then the error should be made to sound least objectionable to the ear.

These subjective considerations appear to offer challenging areas for further research into the problems of sound recording.

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