A MONAURAL DIPLACUSIS AUDIOMETER

by

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A Thesis Submitted to the Faculty of the

DEPARTMENT OF ELECTRICAL ENGINEERING

In Partial Fulfillment of the Requirements
For the Degree of

MASTER OF SCIENCE

In the Graduate College

THE UNIVERSITY OF ARIZONA

1962

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ABSTRACT

This paper presents a theory for an explanation of monaural diplacusis. The theory is supplemented by the behavior of an electrical analog of the human cochlea. Design of a monaural diplacusis audiometer is detailed. Experimental results of tests for six people are presented and compared with theory. Because of the excellent comparison, experimental results serve to strengthen the underlying pattern-place theory for hearing and recognition.

ACKNOWLEDGMENTS

The author wishes to gratefully thank Dr. J. L. Stewart for his many helpful suggestions and his invaluable aid which made this paper possible. Also the author wishes to thank E. Glaesser and W. F. Caldwell for help in using the electrical analog cochlea for diplacusis prediction. The author is also indebted to the several subjects, and especially Pete Wagner, who willingly gave of their time for testing purposes.

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CHAPTER ONE

THEORETICAL DEVELOPMENT

1.1 Introduction

Monaural diplacusis which is associated with nerve damage in the inner ear is identified by the audiologist when a pure tone, presented unilaterally, is reportedly heard by a subject as a noisy, somewhat random tone. In the case of extremely localized nerve damage, a single tone may be heard as two entirely separate tones.

We will redefine monaural diplacusis as the apparent change in pitch (pitch being the subjective measure of frequency) of a tone partially masked by noise as compared to the same tone unmasked by noise. This definition covers a much broader range of nerve damage than the previous definition since it need not imply gross trauma to produce a detectable pitch change.

Diplacusis may be produced by permanent nerve damage as in presbyacusis, lesions along the basilar membrane, congenital defects, or any action that will produce a non-reversible loss in sensory discrimination, or it may only be temporary as in fatigue produced by exposure to high but not traumatic sound intensities.

It is the purpose of this paper to show that this diplacusis is indeed related to nerve damage in the cochlea, and hence to hearing loss. Also the extent and regions of this damage may be identified in terms of particular positions along the basilar membrane.

1.2 Physiological Background

A sound impinging on the pinna passes through the outer and middle ear and into the inner ear. The inner ear is called the cochlea and is composed of the scala vestibuli and the scala tympani (both fluid filled channels) separated by the cochlear duct. On the basilar membrane of the cochlear duct is the Organ of Corti with its associated hair cells, and overlying this is the tectorial membrane. Figure 1.1 shows a cross section of the cochlea.

As a result of sound stimulation, a vibrational pattern is set up along the basilar membrane such that for each different sustained sound there arises a definitive pattern.

It is believed that the relative motion between the tectorial membrane causes a shearing action on the cilia of the hair cells of the Organ of Corti. This action results in a "firing" of the hair cells. Since the shearing force on the cilia is produced by the motion of the membrane, the position of the excited hair cells and the rate at which they fire depend on the vibrational pattern of the membrane.

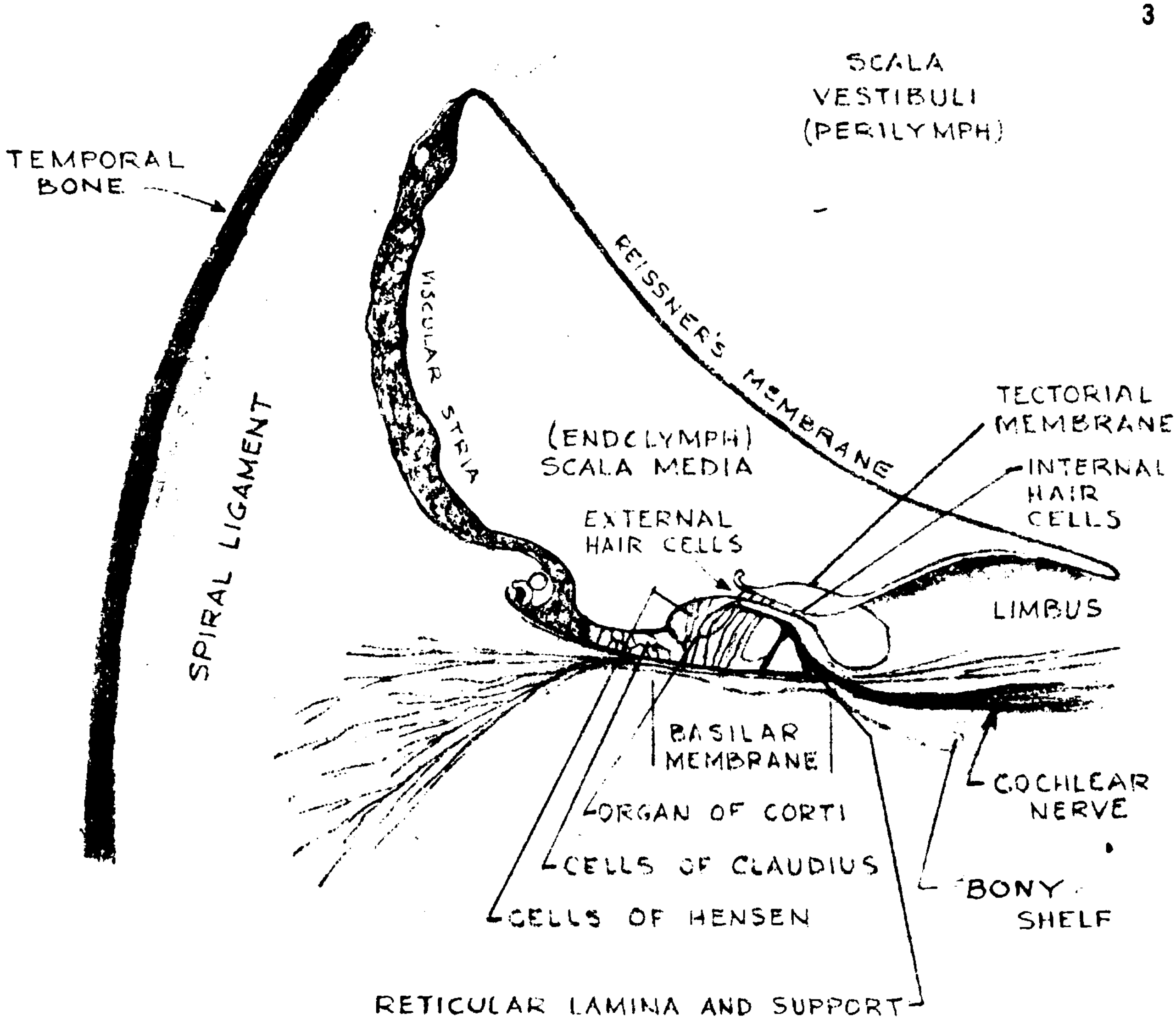

FIGURE 1.1 CROSS-SECTION OF THE COCHLEAR DUCT

The rate of firing, at least for the higher frequencies, is in direct relation to the intensity of the stimulus and is believed to represent intensity in the neural representation.[1] This relationship does not hold at high intensity levels due to the refractory time of individual hair cells. As intensity is increased, each cell ultimately reaches its maximum firing rate and further increases in intensity no longer produce more rapid firing. The neural pattern is then no longer a true representation of the stimulus and the ear is said to have reached saturation.

The pattern of the neural impulses pertaining to hair cells along the basilar membrane is believed to rather closely approximate the membrane motion. That is, the hair cells of the Organ of Corti act as mechanical-electrical transducers converting the membrane motion into neural form which then travels to the brain.

1.3 Theory of Pattern Recognition

At this juncture one may well ask what feature it is in the neural equivalent pattern, or equivalent vibrational pattern, that is important to the brain for recognition. There are a number of theories in answer, or partial answer, to this question. One of the more logical of these is the "place" theory of hearing. This theory contends that position along the basilar membrane of some feature of the vibrational pattern is of major importance to recognition, or at least to pitch determination. This feature is usually taken to be the position of the point of maximum vibration, but just as valid a specification might be the centroid of the pattern. Although cases can be made for other, non-ambiguous characteristic features of the pattern, we will assume that it is the centroid of the pattern shape that is of prime importance.

Although the place theory is widely accepted and can explain many psychoacoustic phenomena, it is still somewhat incomplete. To augment the place theory we supplement it with the pattern theory. This theory contends that it is the pattern's shape that the brain utilizes for recognition. This may be partially true but it has been noted that pattern shapes are much the same for a large range of sine wave frequencies and indeed for other more complex sounds.[2,3] Reasonably the discrimination present in the ear could not be accounted for solely on the basis of slight pattern shape changes.

We therefore hold that both the pattern shape and some distinctive feature of this shape, (e.g., the centroid), are used in recognition. These two properties may not be of equal importance in all cases, or even in most cases, but both are believed to be important to recognition in general and recognition of a tone in particular.

As stated before, the neural equivalent pattern is sent to the brain for the recognition process. It is reasonable to hypothesize that the cortex performs a sort of correlation between the data sent to it from the ear and data stored in the memory. This memory should consist of a file of patterns built up over a period of time as a result of experience and learning. Thus, the first time we are exposed to a noise or signal, we may fail to achieve recognition, or achieve an erroneous identification of the signal, simply because there is no equivalent pattern in our memory. We may get an erroneous identification by virtue of the fact that, if there is a somewhat similar pattern in our memory, the mind may choose an identification that results in the closest fit to the input signal in some sense, as possibly least mean-square error.

How our mind is able to accomplish correlation and the specific regions that do it need not concern us here. It is sufficient for us that the brain is able to compare two signals, closely spaced in time, and to discriminate between relatively small differences in pitch between the two. Our long term memory of any specific tone is very poor; we can identify only about seven classifications of pitch if time between comparisons is long. Our short term memory is very much better and fairly accurate judgments may be made in comparing two tones where the shorter the interval between comparisons the more sensitive will be the comparison, up to a point.

The difference limen is a measure of just how accurately the ear may distinguish differences in pitch between two tones of the same intensity. The difference limen, or dl, is defined as the smallest difference in frequency we may establish and still discriminate between two different tones.

Table 1 gives the values of the difference limen for various frequencies for two intensities of sine waves,[4] which are the two intensities used in tests to be described in this report. It may be noted that, as frequency is increased, a larger change is required in order to be detected.

There is also evidence for a subjective change in pitch between two tones of the same frequency which have different intensities.[5] This may be explained if the ear is a power law detector with a power law exponent somewhat larger than unity and if the pattern is asymmetric. Figure 1.2 shows a typical pattern along the membrane for a high frequency tone. Figure 1.2 also shows another pattern for the same tone but at a higher intensity. If the power law exponent is slightly greater than unity, it is readily apparent that the pattern shape is changed and hence subjective identification is modified. The pattern shift is theoretically toward higher frequencies with higher intensity and this is borne out by experiment at high audio frequencies.[4] At lower audio frequencies, by virtue of a change in the characteristic pattern shape, the shift is toward lower frequencies with increased intensity.

Table 1

Difference Limen

Frequency (cps)	Difference Limen (cps)	
	15 db above threshold	25 db above threshold
1000	5.0	4.0
1500	6.0	4.5
2000	8.0	6.0
3000	12.0	9.0
4000	16.0	12.0
6000	27.0	20.0
8000	38.4	32.0

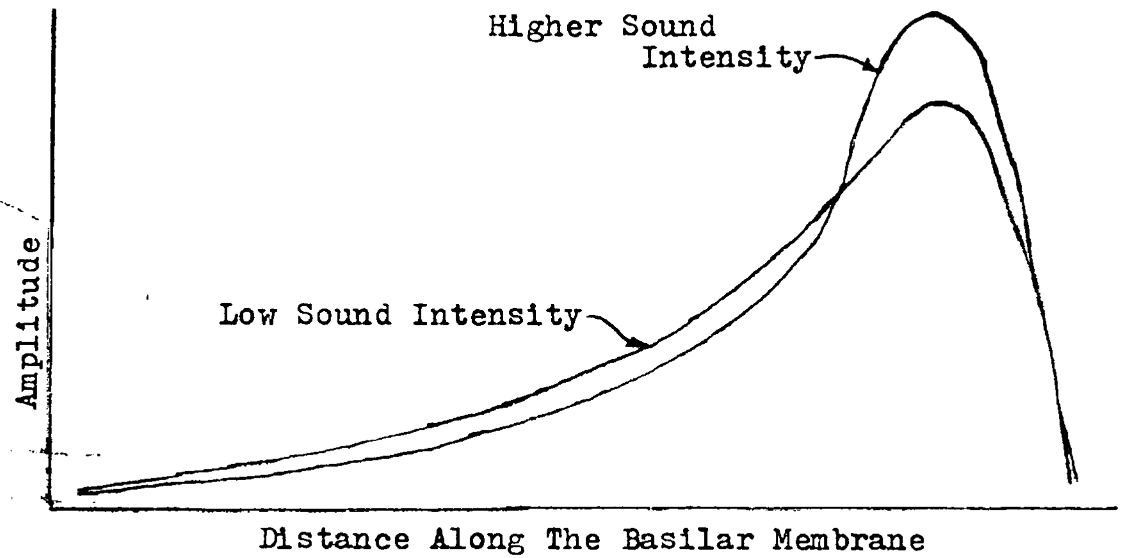

Figure 1.2 Neural Pattern Along the Basilar Membrane for Two Sound Intensities.

1.4 Development of the Recognition Function

It would be very informative at this point to develop the relationship between subjective loudness and the time-pressure waveform of the sound. There is sufficient experimental evidence available to suggest a relationship as

$$L = KF^{n/2} \tag{1}$$

where K is a constant of proportionality that may be related to the individual ear, n is an exponent (approximately unity) and F is the mean-square value of the stimulus temporal waveform f(t).

In generalized equation form we presume that, for f(t) small

$$\begin{aligned} L &= \mathrm{Av} \sum_{k=0}^{\infty} C_k \left| f(t) \right|^{n+k} \\ &= \mathrm{Av} \left[C_o \left| f(t) \right|^{n} + C_1 \left| f(t) \right|^{n+1} + \ldots \right] \\ &\simeq \mathrm{Av} \left[C_o \left| f(t) \right|^{n} \right] \end{aligned} \tag{2}$$

in which "Av" denotes a suitable time average. For convenience C_o is assumed to be unity in the subsequent development, which also presumes that C_o is independent of time.

For infinite-duration averaging without time weighting equation 2 becomes

$$L = \lim_{T \to \infty} \frac{1}{T} \int_{-T/2}^{T/2} \left| f(t) \right|^{n} dt \tag{3}$$

which is familiar in communications theory.[6]

The use of the absolute value of the gross stimulus f(t) implies rectification of the function; the ear is known to perform this operation.[7] Although this equation is for infinite time averaging, an obviously impossible physiological process, it is valid insofar as the averaging time is long in comparison with the periodicities of the waveform.

To be specific we may let the time function be composed of an information signal s(t) and a noise signal n(t) as

$$f(t) = s(t) + n(t) \tag{4}$$

For our development we take the relatively simple case of s(t) a sine wave and n(t) a Gaussian random noise variable. We will make the assumption that these are statistically independent and adopt the convention that N is the mean square value of the noise and S is the mean square value of the sine wave. Then

$$F = N + S \tag{5}$$

We hereafter assume full-wave rectification, although results are the same as for half-wave rectification.

Substitution of equation 5 in equation 3 results in, after a rather complicated integration, the loudness function we desire as,[7]

$$L(S, N, n) = \frac{\Gamma(1+n)}{\Gamma(1+n/2)} \left(\frac{N}{2}\right)^{n/2} {}_1F_1\left(\frac{-n}{2}, 1, -S/N\right) \tag{6}$$

where ${}_1F_1(-n/2, 1, -S/N)$ is the confluent hypergeometric function.

A series representation for equation 6 will result if we expand this function as

$$ {}_1F_1(\alpha, \beta, x) = 1 + \frac{\alpha}{\beta}\frac{x}{1!} + \frac{\alpha}{\beta}\frac{(\alpha+1)}{(\beta+1)}\frac{x^2}{2!} + \dots $$

$$ = \sum_{n=0}^{\infty} \frac{\alpha_n}{\beta_n}\frac{x^n}{n!} \tag{7} $$

where $\alpha_n = \alpha(\alpha+1)(\alpha+2)\dots\dots(\alpha+n)$

and $\beta_n = \beta(\beta+1)(\beta+2)\dots\dots(\beta+n)$

Substituting our values in the equation results in

$$ {}_1F_1\left(\frac{-n}{2}, 1, -S/N\right) = 1 + \frac{n/2(S/N)}{1!} + \frac{n/2(n/2+1)}{2!}\frac{(S/N)^2}{2!} + \dots \tag{8} $$

Our loudness function now becomes

$$ L(S, N, n) = \frac{\Gamma(1+n)}{\Gamma(1+n/2)}\frac{(N)^{n/2}}{(2)^{n/2}}\left[1 + \frac{\frac{n}{2}\left(\frac{S}{N}\right)}{1} + \dots\right] \tag{9} $$

It is claimed that , with good accuracy, we may represent a sine wave in noise in terms of a relatively narrow band of noise in a broader background of noise (i.e., $N = N' + S'$, where S' is the narrow band of Gaussian noise and N' is the broad band of noise).[7] If we now let the mean square value of the sine wave be equal to zero in equation 6 and use the above representation for N we achieve a loudness function

$$ L(0, N, n) = \frac{\Gamma(1+n)}{\Gamma(1+n/2)}\left(\frac{N'+S'}{2}\right)^{n/2} \tag{10} $$

since the hypergeometric function reduces to unity. This equation will assume importance subsequently in the development of a recognition function.

For N>>S, equation 9 becomes

$$L(S, N, n) \simeq K\ N^{n/2} \tag{11}$$

This means that, at positions along the basilar membrane where the noise is relatively intense, the neural equivalent is almost entirely the result of the noise component of the signal.

On the other hand, for S>>N we obtain from equation 8

$$L(S, N, n) \simeq K'S^{n/2} \tag{12}$$

This result indicates that, where the information signal is more intense than the noise, the resultant pattern is due almost entirely to the signal.

An interesting sidelight of equations 11 and 12 is that, for n equal to unity, the loudness functions are proportional to the rms values of the noise and the sine wave respectively.

To show the validity of equations 11 and 12, pictures were taken of the response of an electrical analog of the human ear to noise, signal and combined signal and noise inputs.[8]

Figure 1.3a is the response to a Gaussian noise input where the ordinate represents amplitude and the abcissa distance along the basilar membrane. The response is not flat as one might first expect because

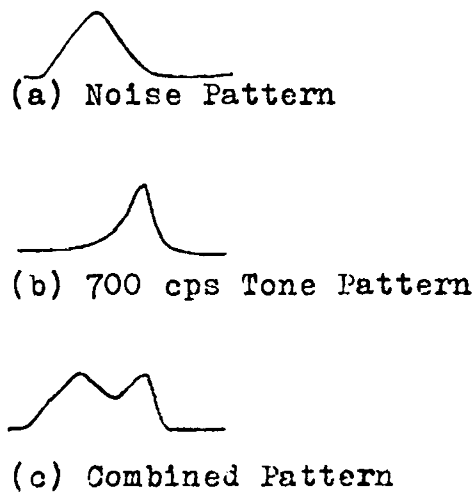

Figure 1.3 Analog Cochlea Patterns for a Normal Ear.

the ear is not equally sensitive to all frequencies. Thus a spectrum in which all frequencies are equally represented ought to evoke a response that reflects essentially the shape of the threshold curve of the ear. Wever has shown that, if we properly shape the noise, we get essentially a uniform response in the ear.[1] His shaping curve (obtained by experiment) is approximately the inverse of the curve in Figure 3a.

Figure 1.3b is the pattern for a tone of 700 cps and Figure 1.3c is the curve for the sum of noise and signal. As may readily be seen, this pattern shows the foregoing conclusion that the resultant is dependent on the relative signal to noise levels at each individual point along the membrane. The analog ear has power law detector exponent of approximately unity and hence the curves represent the rms values of the input.

It may be noted that the pattern for the 700 cycle tone is asymmetrical. This is because the basilar membrane vibrates in a very characteristic manner with frequency. At low frequencies the membrane tends to vibrate as a whole with the maximum vibration at or near the helicotrema. As the frequency is increased the position of maximum vibration moves toward the stapes. The entire membrane preceding the maximum shows some motion but this motion is heavily damped further along the membrane.[2] Since the firing of a hair cell depends on the motion of the membrane, neural pulses reflect this asymmetry. As intensity is increased the cell responds to a wider and wider range of frequencies around the frequency of maximum response. The neural equivalent reflects this as is shown by experimental measurements of cochlear potentials.[4]

When noise is added to the tone, recognition becomes more difficult since the noise tends to mask the tone; the result is in part a subjective decrease in loudness of the tone. The addition of noise may also change the shape of the neural equivalent pattern and subsequent recognition may be modified.

In order for recognition to take place there must be a definable recognition function. Logically this is the difference between the loudness function for signal plus noise and that with only noise present.[7,9] In equation form, this becomes

$$\begin{aligned} R(0, N, n) &= L(0, S' + N', n) - L(0, N', n) \\ &= K\left[(S' + N')^{n/2} - (N')^{n/2}\right] \end{aligned} \tag{13}$$

The relationship of equation 13 might be what the central nervous system uses for comparison with the memory patterns in order to achieve recognition.

For a person with nerve damage, the neural pattern can obviously not be the same as for an undamaged ear. Nevertheless there must be some type of identification process since people with damaged ears can still hear tones, however imperfectly. We learn with experience to identify a given sound with a particular pattern shape, and if something changes this shape we must change our identification correspondingly.

1.5 Diplacusis Prediction Using an Analog Cochlea

We will try to predict what may happen in a damaged ear with respect to recognition. For this we use the analog ear in which the frequency response from 1.5 kc to 3.5 kc has been deleted with some response lost over an even wider range. The specific trauma constitutes total nerve destruction over a limited region, while the rest of the cochlea remains normal. (This type of destruction can result from exposure to pure tones at high intensity levels.[2])

Figure 1.4a is a tracing of a photograph of the response of the damaged analog ear to noise which may be compared to Figure 1.3a for the healthy ear. Figure 1.4b is the pattern for a 1 kc tone, which is at a frequency below the damaged area. Figure 1.4c is the resultant of noise and the 1 kc tone. We may now perform our suggested recognition function on the resultant. This can be done with a graphical procedure as shown in Figure 1.4d. Comparison of Figures 1.4b and d shows that the centroid of the pattern has been shifted to the left, which corresponds to a lower frequency, although the pattern shape is not much altered.

If the tonal frequency is above the damage, then when noise is added and the recognition function is obtained, a resultant shift upward in frequency is found. Figures 1.5a, b, and c show this in the same

(a) Noise Pattern

(b) 1 kc Tone Pattern

(c) Combined Pattern

(d) "Recognized" Pattern

Figure 1.4 Analog Cochlea Patterns for an Ear With Midfrequency Damage--Tone Below the Damage.

(a) 5 kc Tone Pattern

(b) Combined Pattern

(c) "Recognized" Pattern

Figure 1.5 Analog Cochlea Patterns for an Ear With Midfrequency Damage--Tone Above the Damage sequence as before, as pure tone, resultant, and recognized pattern for a tone of 5 kc. The noise pattern is the same as in the previous example.

Perhaps a more common type impairment is the loss of sensitivity to higher frequencies, as in presbyacusis or in some forms of injury caused by illness. This form of damage is usually characterized by little or no low frequency loss and by an increasing loss with increasing frequency.

In order to simulate this type of damage on the cochlea, the gains of the various sections of the analog of the cochlea were adjusted to give a linearly increasing loss on a logarithmic basis starting at 3 kc and extending to 10 kc; after this point the loss was maintained constant at about 90 per cent.

Figure 1.6a is the resulting pattern due only to noise. Comparison with the previous noise (for a damaged region at midfrequencies) shows considerable difference. The cochlea was driven rather intensely in order to achieve the result of Figure 6a; what was once only a small portion of the total noise pattern now constitutes almost the entire pattern.

The pure tone pattern is also considerably altered. There is so little high frequency response that, rather than the characteristic asymmetrical pattern shape, the pattern is almost symmetrical. It is

(a) Noise Pattern

(b) 3 kc Tone Pattern

(c) Combined Pattern

(d) "Recognized" Pattern

Figure 1.6 Analóg Cochlea Patterns Characteristic of Presbyacusis--Midfrequency Tone.

thus seen that one of the clues to normal recognition may have been lost in the associated neural process, namely that of pattern shape.

Figure 1.6c results from combined signal and noise and 1.6d shows the pattern which results upon applying the graphical recognition function where, superimposed on this pattern, is the original signal pattern. In addition to less signal intensity because of normal masking, there is also a centroid shift. The shift is toward the right, which corresponds to a shift toward a higher frequency.

Shown in Figure 1.7 without explanation are the various patterns for a 10 kc tone under identical noise and damage conditions as for Figures 1.6. Again the shift is toward a higher frequency.

One may ask why in this case the diplacusis is always upward while in the previous case of midfrequency damage the diplacusis depended upon the test frequency in relation to the position of maximum loss. A close study of the noise pattern shape is informative.

In order to achieve approximately the same relative masking in the second case as in the first, considerably more low frequency noise is required which effectively shifts the noise pattern along the basilar membrane toward the helicotrema. This means that, for tonal frequencies in the damaged area, the noise envelope is largely to the left of the signal pattern. Therefore the recognized pattern is shifted to the right.

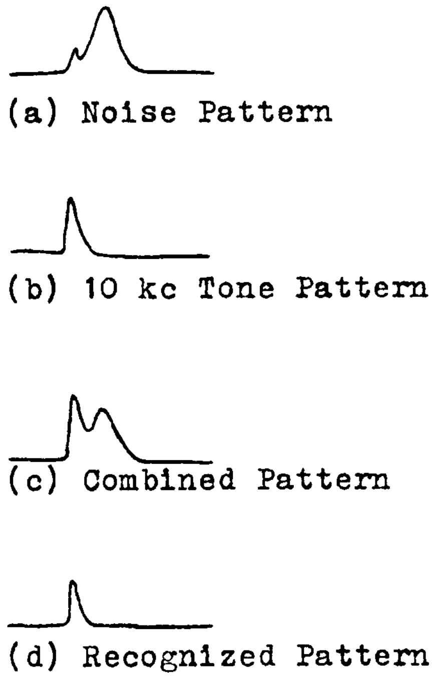

Figure 1.7 Analog Cochlea Patterns Characteristic of Presbyacusis--High Frequency Tone.

In the first case with localized damage the pattern is relatively unchanged and hence a good deal of the noise pattern is to the right of the signal pattern when the tone is lower than the damage. Hence the diplacusis is downward. When the tone is above the damaged region, the situation is essentially reversed and so the diplacusis shifts to above the reference tone.

1.6 Summary of Theory of Diplacusis

We have tried to show that the response of a hair cell depends upon its place along the membrane and the intensity of the vibration at that point. The net effect is that individual hair cells act as if they were somewhat selectively tuned to respond more readily to certain frequencies than to others, and the frequency of the maximum response is a function of the position of the hair cell along the membrane.

The resultant neural pattern constitutes a reasonably accurate representation of the motion of the basilar membrane and it is this neural equivalent pattern that the brain uses for recognition. If there occurs a change in the characteristic pattern shape for a given sound, as certainly occurs with nerve damage, there will arise complications in pattern recognition until a memory can be "organized" for this distorted pattern. The brain, in performing recognition with stored information, is forced to work with imperfect data such that a slight change in the neural pattern may produce recognition error. If the change is gross, normal recognition can not take place and all we hear is noise.

It may also be hypothesized that the information sent to the cortex contains more clues to recognition than are actually required. It is then still possible to achieve correct identification even though one or more of the clues is not present in the neural signal or is somehow lost in transmission. Loss of some number of the totality of clues will in any event result in a pattern which does not promote correct identification. Of course, a change in any of the clues present in the pattern makes correct identification more difficult.

Anything that produces change in the pattern, such as noise-induced distortion (e.g., partial masking), will change the recognition function. It is one aspect of noise-induced change that we call monaural diplacusis. If our assumptions are correct, the more imperfect is the neural equivalent pattern, that is, the more extensive is the nerve damage, the greater will be the subsequent change with noise addition. Because the changed neural pattern is recognized as a tone of a different pitch, we have a measure of nerve damage in the pertinent region of the cochlea.

CHAPTER TWO

DIPLACUSIS AUDIOMETER DESIGN

The diplacusis audiometer consists of two major parts: an audio frequency sine wave generator and a white noise source. Both basic circuits are straightforward.

2.1 Audio Frequency Tone Generator

The main requirements for the audio oscillator are that it have good stability and that it have adequately pure sine wave output. One wave-form generating circuit that has these features is the modified Wein bridge oscillator.[12] In this circuit, a two stage "flat" amplifier is enclosed in a frequency selective feedback network with positive feedback from the plate of the second stage to the grid of the first stage. Due to the extreme frequency selectivity of the bridge circuit in the feedback loop, there is high rejection of undesired signal frequencies compared with the selected frequency. The result is a very pure sine wave of the desired frequency as long as feedback is limited so that the circuit is not overdriven and distortion introduced.

An added attraction to this type of circuit is that it readily lends itself to the type of modification required for the present purpose.

The active circuit used here is basically that of an Eico model 377 audio generator. This circuit provides nearly constant amplitude-frequency characteristics with good frequency stability. The inclusion of variable negative feedback increases the stability of the gain of the oscillator circuit. Incorporated in the cathode of the first stage of the oscillator is a resistor with a positive temperature coefficient. This produces cathode degeneration to keep the tube operating in its linear characteristic region. It also helps to maintain a constant output amplitude and as such constitutes an alternate to standard AVC techniques.

If the resistors in both branches of the feedback loop are equal and the two capacitor values are likewise equal, the frequency determined by the feedback network is proportional to the reciprocal of the product of resistance and capacitance. Values chosen are such that there is realized continuous frequency variation from 700 cps to well above the normal audio range.

The feedback capacitance consists of two separate capacitances in parallel. The larger ganged capacitor is always in the loop; the smaller variable ganged capacitor can be switched out of the circuit and replaced with a fixed padding capacitor. The padding capacitors can be adjusted so that, when the smaller ganged capacitor is set at its mid-value, switching between the fixed and variable capacitors produces no shift in frequency. Thus when the padding capacitors are in the circuit a frequency is obtained that may be interpreted to be the standard or reference tone; this frequency is determined only by the setting of the larger ganged capacitor. In the other switch position, the smaller capacitor can continuously vary the frequency for a considerable range below and above the reference frequency. With this simple modification, there is obtained a variable tone (around a preselected reference) for easy and rapid comparison to the reference tone.

A cathode-follower is used for the output stage. The desired frequency is fed from the two stage oscillator circuit, through a variable gain control, to the cathode-follower. This stage is required when the device is used with a low impedance headphone set.

It may be noted in the circuit of Figure 2.1 that there is a fixed attenuator in the standard frequency path while in the variable frequency path there is an adjustable attenuator. These attenuators allow a match between the subjective loudness of the two tones to be obtained when noise is superimposed. This maneuver allows for a subjective decrease in the intensity of the tone with the addition of noise -- this prevents a change in intensity from causing a subjective change in pitch.

2.2 The White Noise Source

The noise source as shown in Figure 2.2 is a grounded-grid thyratron in a magnetic field. This type circuit gives the desired flat

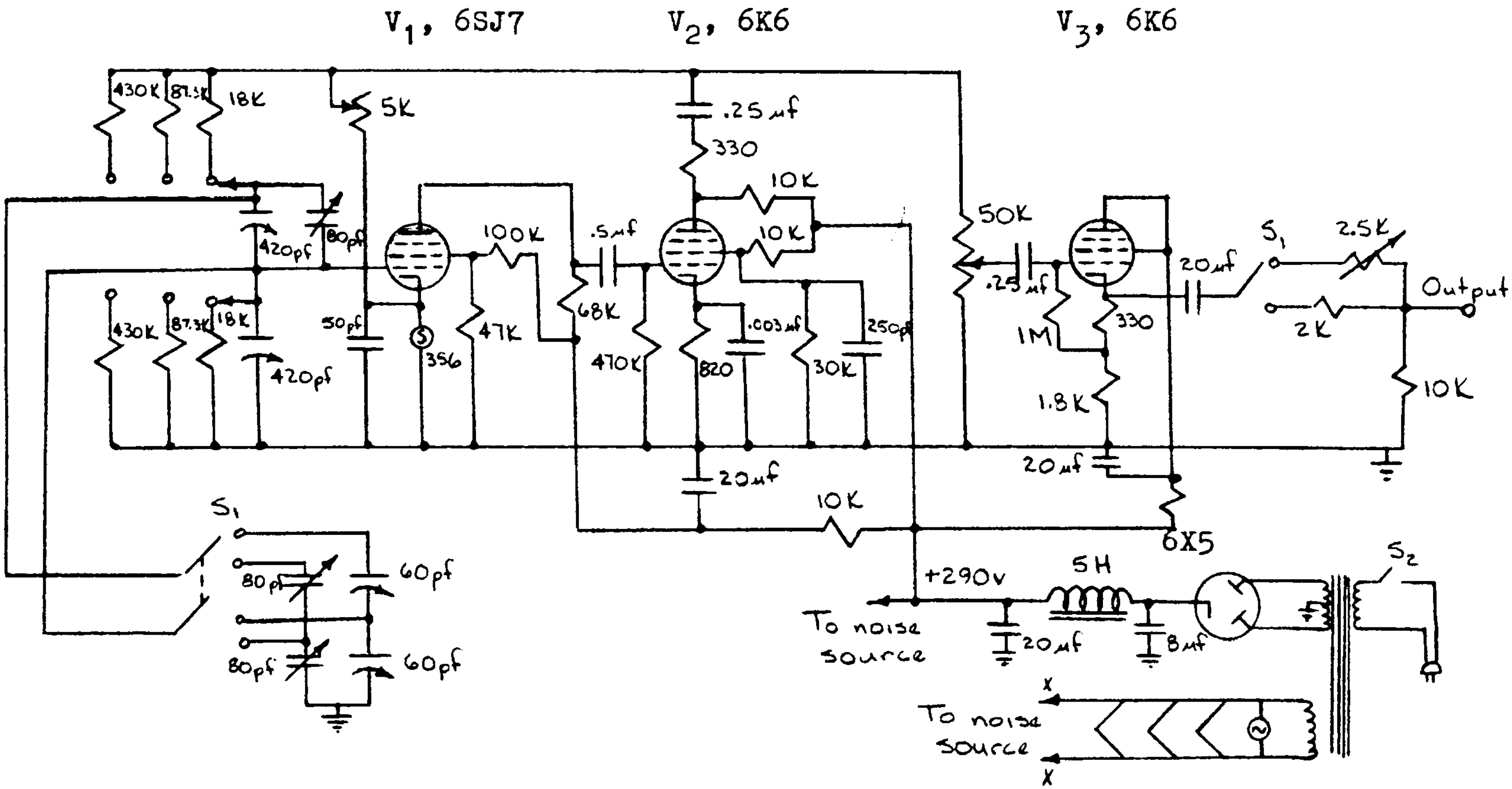

Figure 2.1 Oscillator Circuit and Power Supply.

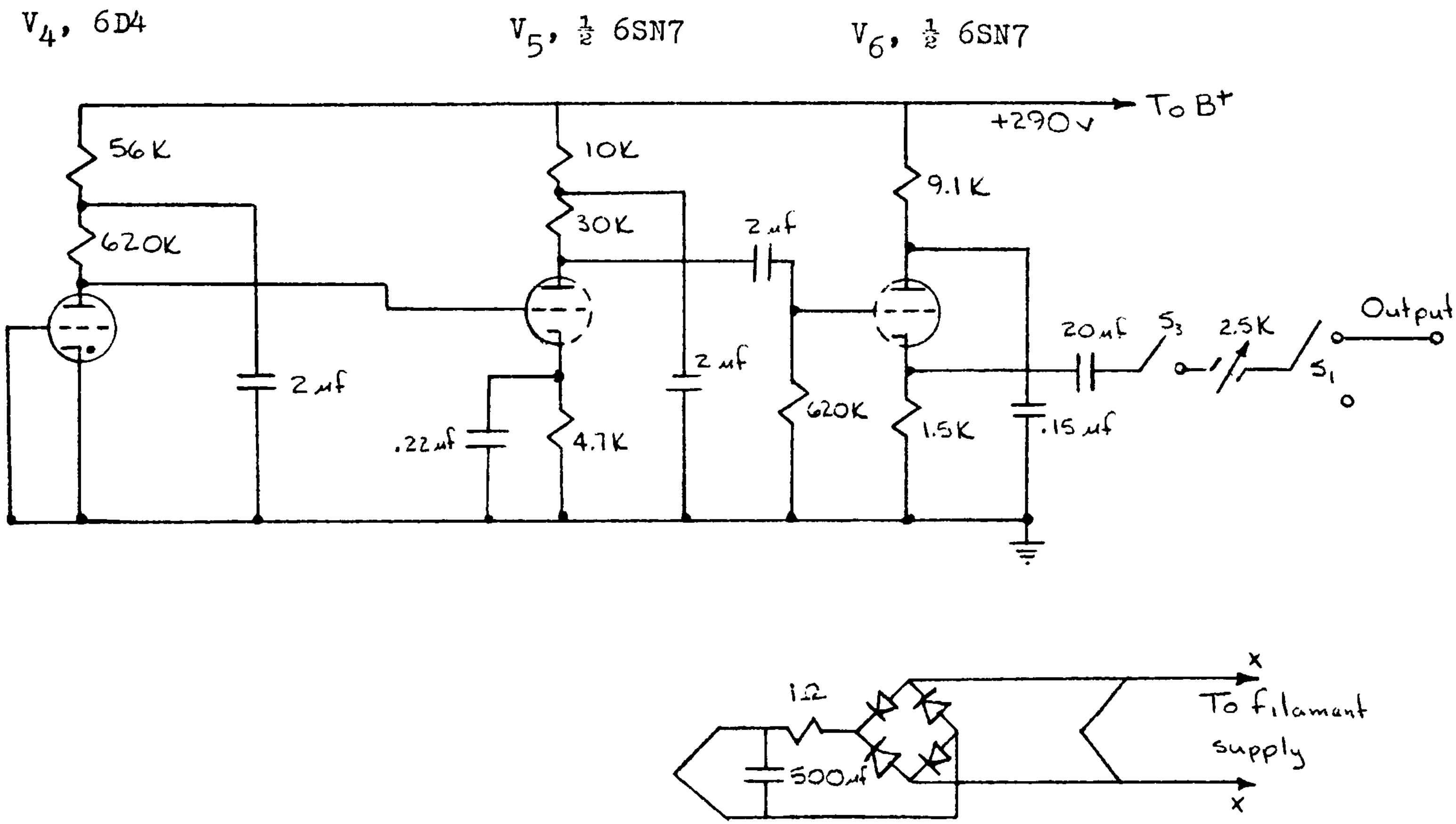

Figure 2.2 Noise Source.

frequency spectrum and is well known.[13] Direct measurements verify a flat amplitude-frequency characteristic over the range 100 cps to 16 kcps. The magnetic field supresses resonate peaks at frequencies corresponding to natural oscillation frequencies of the gas ions.[13] The magnetic field also increases the generated noise level.

The noise signal is nevertheless relatively low level (.27 volts rms at the thyratron plate) and, since the signal is subjected to considerable gain in a later stage, extra precautions are taken to reduce any 60 cycle or harmonic pick-up. An extra section of RC filtering in the power supply for the thyratron insures negligible 120 cycle hum in the plate supply. (This filter also prevents generated noise from appearing in the oscillator circuit through power supply feedback.) The major aid in minimizing hum is realized through use of dc for the thyratron filament. A separate rectifier-filter combination supplies the required 6 volts dc.

The noise signal is direct-coupled to the amplifier stage. This stage provides a voltage gain of approximately 200 and the signal strength at this point is 4.6 volts rms. An extra section of RC power supply filtering is used with the amplifier for decoupling purposes.

One stage of noise amplification is sufficient and the noise is fed to a cathode-follower output stage. The cathode-follower causes a loss of about 10 per cent and hence the maximum noise output is 4 volts rms, which is sufficient.

After the cathode follower, the signal-noise combination is fed through a variable series attenuator to the switch mentioned in conjunction with the audio oscillator. When the switch is in the position for presenting the variable tone, the noise is also presented. In the other position only the uncorrupted reference tone is presented. Subsequent to the switch, the reference tone suffers a fixed attenuation while the combination variable tone with noise has variable attenuation. By this means, the two tones can be matched in intensity before an attempt is made to match pitch.

In the wiring of the noise source, care must be taken to minimize stray electrostatic coupling. A metallic shield between the noise source and the rest of the chassis is employed and shielded hook-up wire is used for all connections to more distant parts of the chassis. Of course, care is exercised in the layout of the wiring.

2.3 Additional Test Equipment

In cases of extreme hearing loss, the output of the audiometer may be insufficient to provide a signal that is 25 db above threshold. In order to provide sufficient signal strength in these cases a separate amplifier must be used. The amplifier employed in experimentation was a Knight, model 520. Additional work had to be done on this amplifier in order to reduce the hum to an acceptable level. Additional LC filtering was supplied to the power supply and dc filaments were added to several of the high gain stages. The small amount of hum that remained was evidently introduced subsequent to the gain control circuit and it was usually masked by any signal. In any event, attenuators were used following the amplifier and were set so that this small amount of hum was not detectable in the signal presented to the subject.

The attenuator following the amplifier was a Tech Laboratory model 350-A. By means of attenuator settings, signals of known intensity above threshold could be presented to the subject.

For signal presentation to the subject, headphones were used. These were Knight model 840 dynamic phones in which only one of the pair of phones was connected. In this manner the signal could be presented unilaterally while the contralateral ear remained covered so as to seal that ear from external noises which tended to distract the subject.

CHAPTER THREE

TEST PROCEDURE

3.1 Procedure Followed in Diplacusis Testing

In order to measure monaural diplacusis, a subject is asked to match in pitch a partially masked tone with an unmasked tone, both of which are presented unilaterally to the same ear. The test is conducted in the following manner.

First the threshold of the subject is found at each frequency tested. The signal is interrupted by a switch, the gain of the amplifier is varied, and then the tone is reintroduced to the subject. This process is continued until the subject can barely detect the presence of the tone; this defines threshold.

Then, by means of the attenuators calibrated in db, a suitable sound level above threshold is selected, 15 and 25 db in the present case. These values are chosen because they are enough above threshold so that the tones are readily distinguishable and can be matched in pitch but are not so intense that overloading, adaptation, or fatigue become factors in the measurements. In addition to these considerations is the fact that diplacusis tends to disappear at higher intensity levels (which can be predicted on the basis of pattern saturation in the centroid region due to achievement of maximum firing rates of hair cells).[14]

To this signal is added the noise with sufficient amplitude to partially mask the tone. The subject is then required to match in pitch the partially masked tone in one position of his control switch with the pure tone in the other switch position.

The subject can vary the pitch of the partially masked tone by means of a calibrated dial. This dial presents a blank face to the subject and hence gives him no visual clues for picking the "correct" frequency.

With constant comprison to the standard tone, the subject varies the pitch of the masked tone until he judges them to matched in pitch. By bracketing the point of best match a final choice is obtained that is the result of several decisions and thus the variability of the final choice may be reduced.

When the subject announces his final choice the operator reads the calibrated dial and subsequently converts the reading to a frequency value.

This same procedure is followed for each of the frequencies tested. All tests are conducted in a quiet, sound treated room. Results will be discussed in the following section.

CHAPTER FOUR

EXPERIMENTAL RESULTS

On the following pages are presented the results of the diplacusis tests on six persons. In most cases the resultant curve is an average of several tests; different tests of the same individual were so similar that such averaging was permissible.

When the tests were begun it was felt that testing at four frequencies, 1.5 kc, 3 kc, 4 kc, and 5kc, and two intensity levels, 15 and 25 db above threshold, would be sufficient for testing both the audiometer device and the theory. These particular frequencies were chosen since nerve damage usually occurs in this range and matching pitches is not too difficult. As a result of the early testing, it was found that the addition of two higher frequencies, 6 kc and 8 kc, added considerably to determination of the nerve loss. Hence some of the tests shown have only four test points while some of the others have six.

4.1 Diplacusis Measurements in the Normal Ear

In order to demonstrate the practicality of the audiometer, the test was first administered to two people who showed no hearing loss with the standard threshold audiogram. If the theory was correct (assuming no

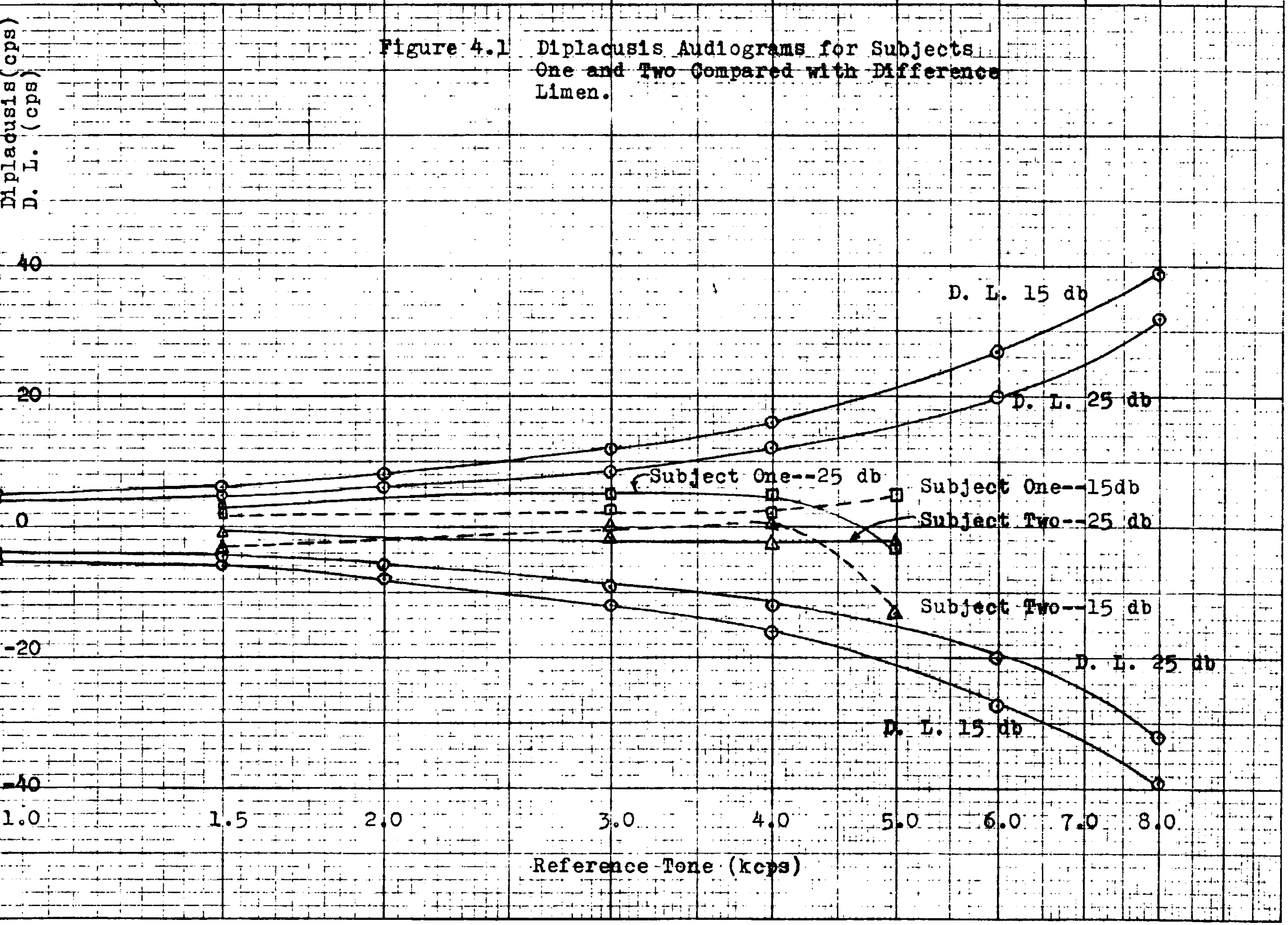

Figure 4.1 Diplacusis Audiograms for Subjects One and Two Compared with Difference Limen.

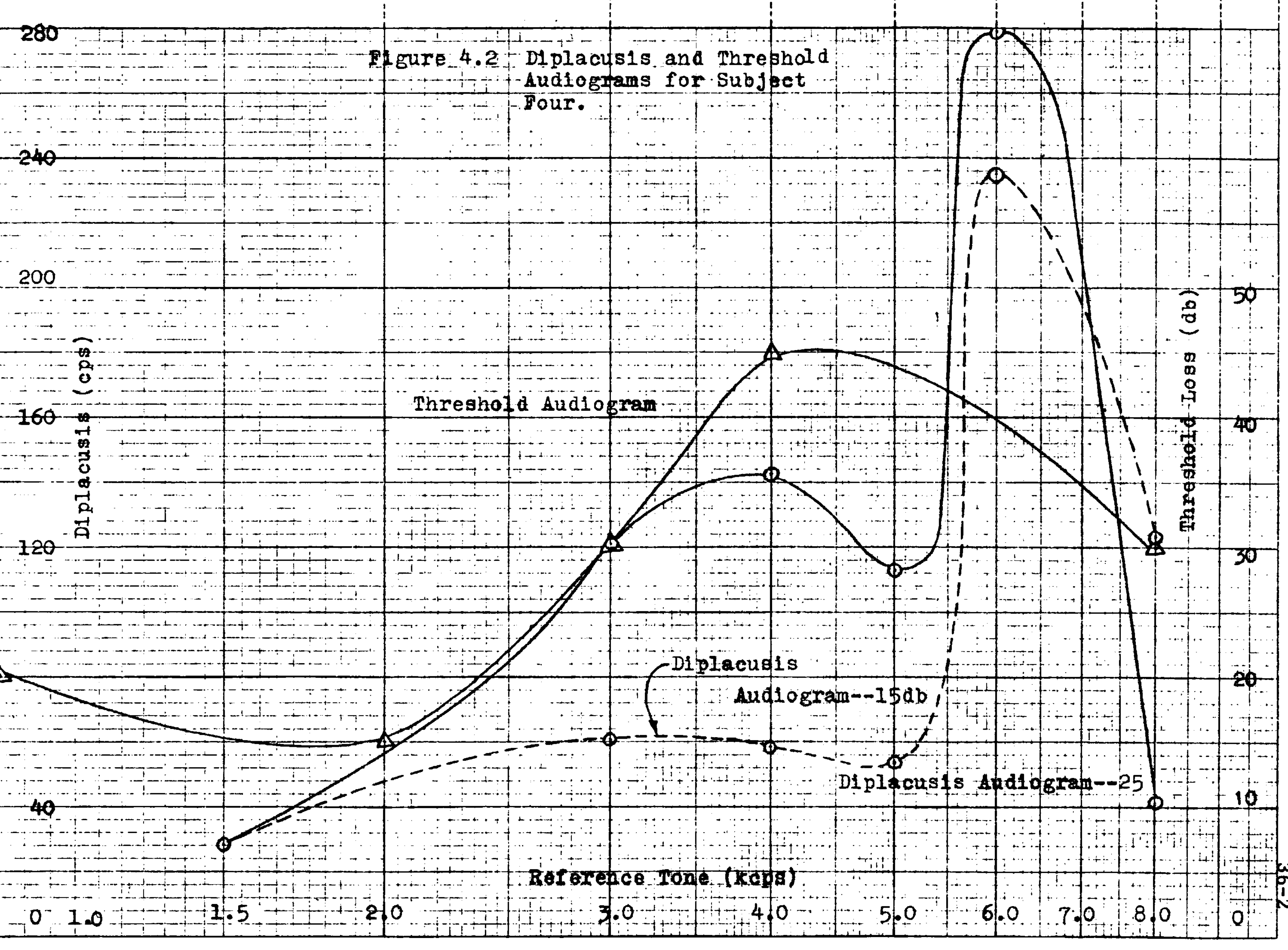
Figure 4.2 Diplacusis and Threshold Audiograms for Subject Four.
280
240
200
160
120
40
Diplacusis (cps)
Threshold Loss (db)
50
40
30
20
10
0
Threshold Audiogram
Diplacusis Audiogram--15db
Diplacusis Audiogram--25
Reference Tone (kcps)
0
1.0
1.5
2.0
3.0
4.0
5.0
6.0
7.0
8.0
36-2

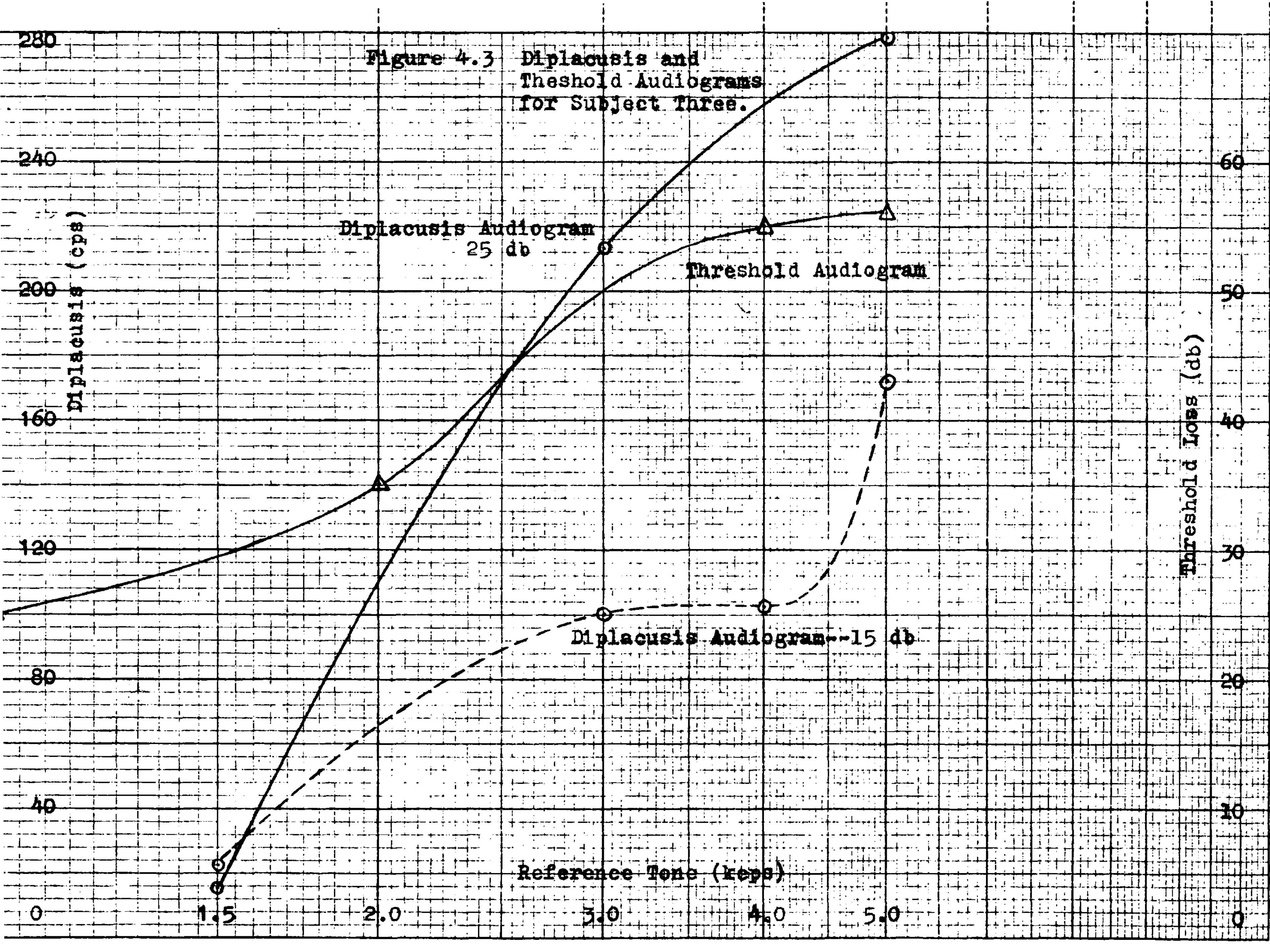
Figure 4.3 Diplacusis and
Theshold Audiograms
for Subject Three.
Diplacusis Audiogram
25 db
Threshold Audiogram
Diplacusis Audiogram--15 db
Diplacusis (cps)
Threshold Loss (db)
Reference Tone (kcps)
280
240
200
160
120
80
40
0
1.5
2.0
3.0
4.0
5.0
60
50
40
30
20
10
0

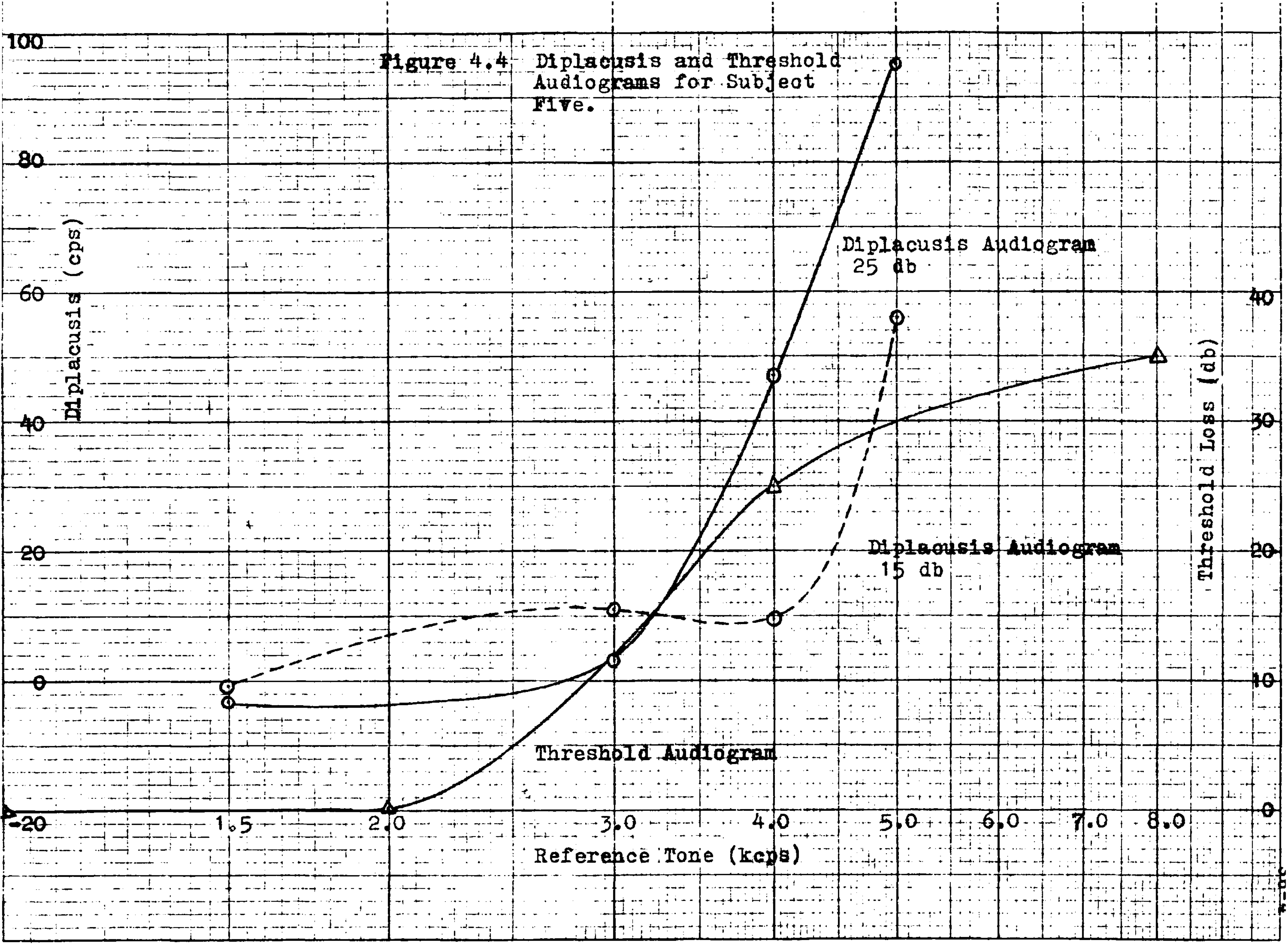

Figure 4.4 Diplacusis and Threshold Audiograms for Subject Five.

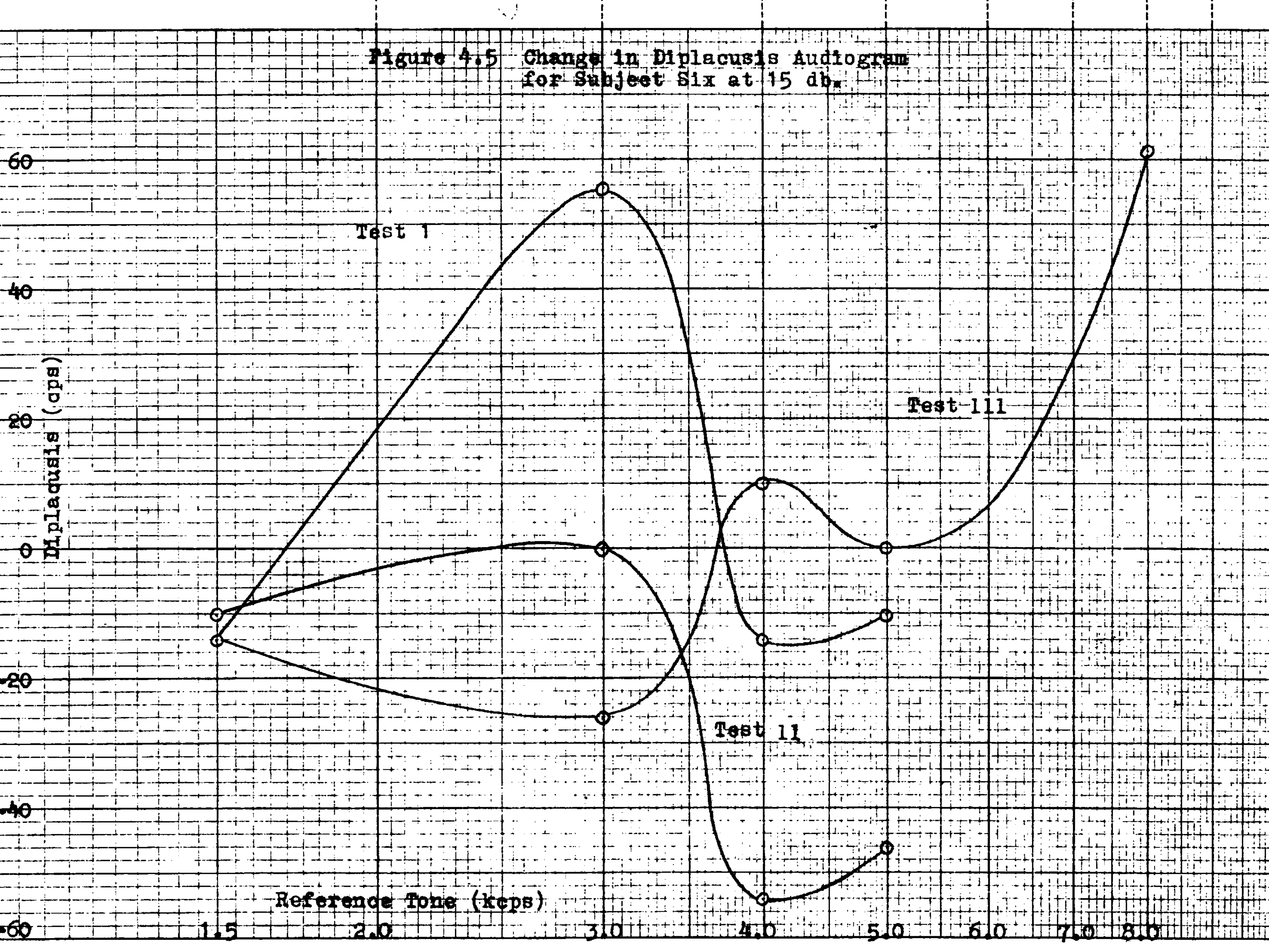

Figure 4.5 Change in Diplacusis Audiogram for Subject Six at 15 db.

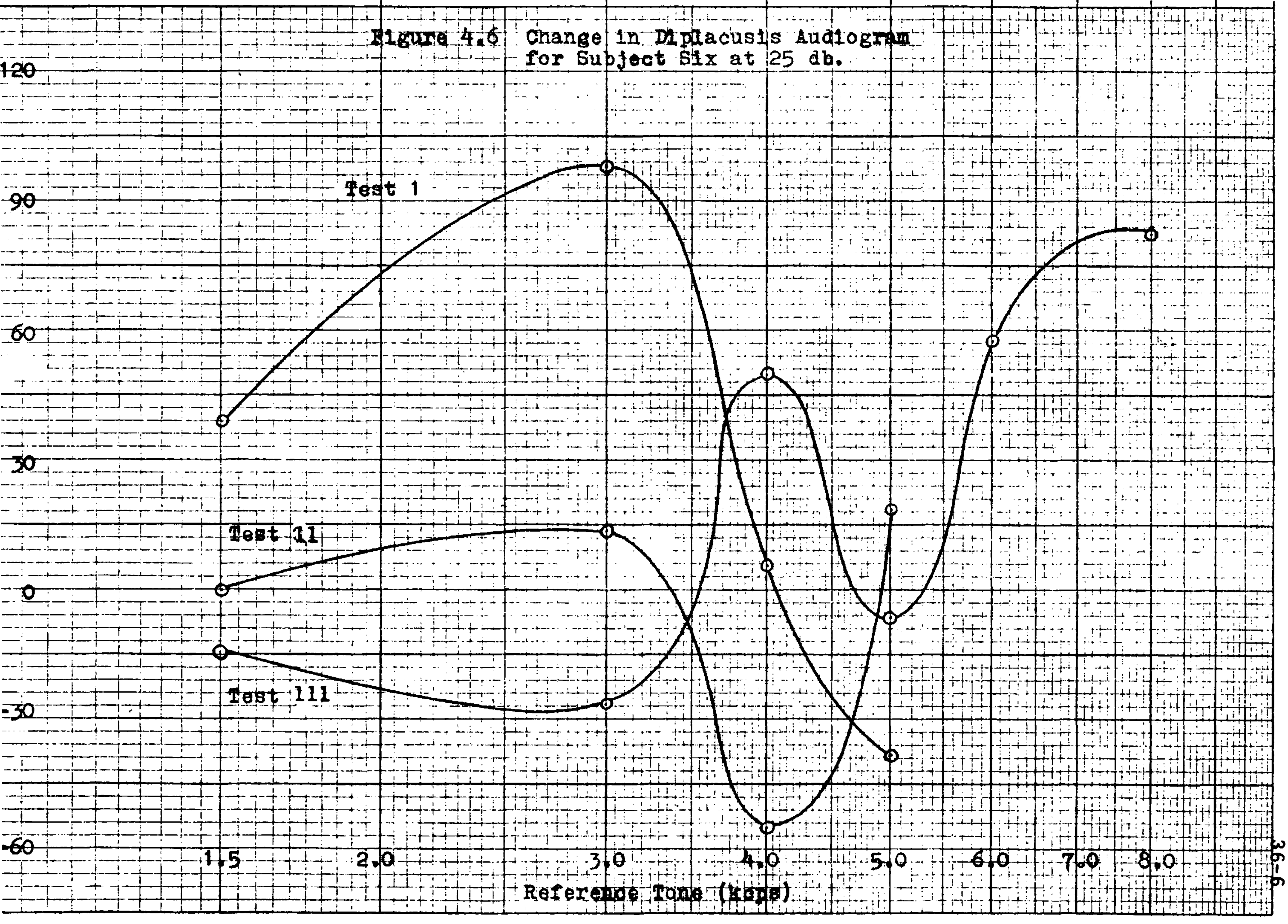

Figure 4.6 Change in Diplacusis Audiogram for Subject Six at 25 db.

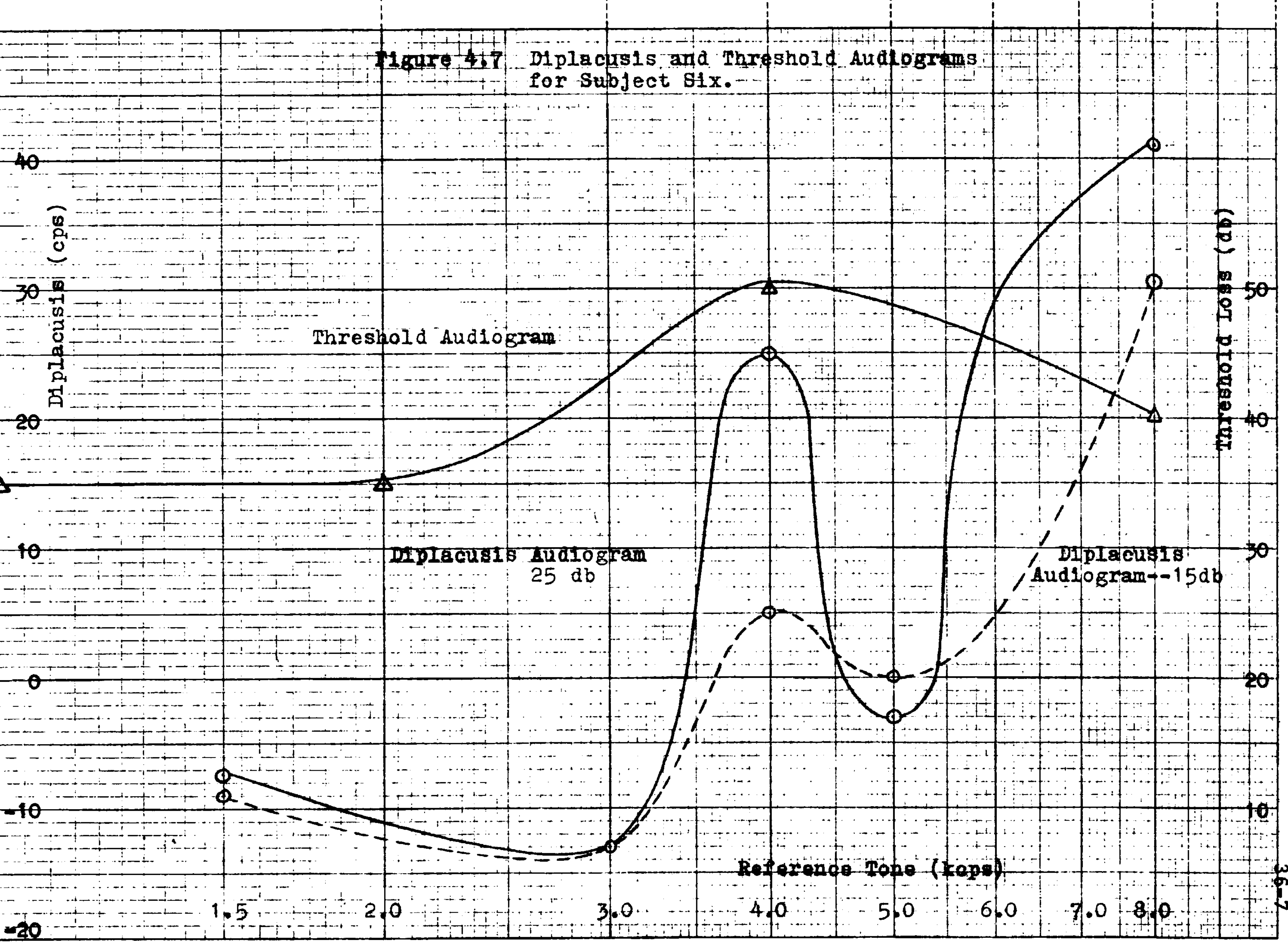

Figure 4.7 Diplacusis and Threshold Audiograms for Subject Six.

major pattern asymmetry for the normal ear) and the audiometer correctly designed, these subjects should not exhibit appreciable diplacusis. Curve 1 applies to two subjects at 15 db intensity above threshold. The bounding curves show the difference limen (where plus or minus uncertainty applies) for the relevant sound intensity. The diplacusis curves give the measured frequency shift and neither curve has values greater than the dl.

Curve 1 also shows results for the same two subjects at 25 db intensity. Again the bounding curves show the relevant dl. The subjects in the tests of curve 1 performed well enough to give some confidence in the design of the audiometer. Also these results demonstrated that a person with no nerve damage shows little or no diplacusis. The threshold audiogram is not also shown on Curve 1 because there was no measurable hearing loss in either subject.

4.2 Diplacusis in the Ear With Nerve Damage

Curves two through four are the results of tests on persons who suffer from varying degrees of presbyacusis, or high frequency hearing loss. Curves of the 15 and 25 db tests for each subject are both shown on the same sheet. Also shown on each curve sheet is the threshold audiogram for the subject.

It is interesting to note how closely the curves for the two intensities (15 and 25 db) are related. They have the same general shape and it is only a shift along the axis which separates them.

The curves for the higher intensity apparently are more regular, probably because, at 25 db, the signal is far enough above threshold to provide better ability in matching the pitch of the two tones. In some cases the subjects commented that it was relatively difficult to distinquish the 15 db tone well enough to be certain of the match. They expressed more confidence in the match at higher intensities although the results for both cases are generally in accord.

The threshold audiograms for the three subjects are also plotted of the respective curve sheets. These curves show the loss of sensitivity of the ear to the selected frequencies due to nerve damage in comparison with what is considered to be "normal."

As postulated in the theory, the amount of frequency shift does indeed appear to bear a direct relationship to nerve damage. It is also evident that, if there is a consistent loss at high frequencies, the diplacusis is upward. In general, the data confirm the hypothesis.

Curve two has one feature worth further comment. This is the fact that the diplacusis audiogram shows less injury at 5 kc and more at 6 kc than is implied by the threshold audiogram.

We know that nerve injury is sometimes non-uniform in the cochlea. That is, there may be patches of almost normal hair cells along the membrane in even the most severely damaged ear. Conversely there may also be completely deadened patches among otherwise uniformly and partially damaged areas. The threshold audiogram is relatively insensitive to such minor variations.

On the other hand diplacusis is rather sensitive to these minor variations since even slight irregularities shift the neural equivalent pattern.

It is apparently demonstrated that the diplacusis audiometer is a much more sensitive instrument for indicating nerve damage than is the threshold audiogram.

4.3 A Special Case of Nerve Deafness Diplacusis Measurement

A most interesting case of a change in pattern recognition is shown in curves five and six, both of which are for the same subject (whose availability is now considered to have been a "fortunate accident"). Curve five is for 15 db and six is for 25 db above threshold. This subject is a good case for the argument that recognition is the direct result of experience and that pattern identification may be changed through a learning process.

As a result of childhood illness, the subject of curves 5 and 6 has an extreme loss at about four thousand cycles. Beginning at lower frequencies, the loss increases gradually until it reaches a maximum and then decreases slightly at higher frequencies; in many respects this is much the same as the case in section 1.5 in which pictures were taken on the analog ear.

In the theoretical development it is stated that, in a case such as this, the diplacusis should be downward at frequencies below the damaged region and upward at higher frequencies. When the subject was first tested the results were exactly the reverse of what the theory stated.

A test made several weeks later showed much the same form of diplacusis but to a lesser extent. Our theory would appear to be hard put to explain behavior such as this.

A third test given the same subject produced the results specified by the theory of diplacusis and a fourth test later confirmed the results of the third test.

The four tests are shown on the two sheets as 1, 11, and 111 where 111 is the average of the last two tests. What then had happened to cause a complete reversal in result with this particular subject? An investigation revealed that, shortly before the first test, the subject had purchased a new hearing aid. Previously he had used an aid which had amplified all frequencies equally such that the aid had not changed the pattern on the membrane or the subsequent irregular and damaged neural pattern but only increased its over-all intensity.

With the new aid, a change in the pattern shape was realized. This occurred since the aid was designed to provide amplification only at frequencies where there was loss, and amplification was in proportion to that loss. This meant that, whereas with the old aid the neural pattern was extremely distorted compared to "normal", the neural pattern with the new aid became much the same as the pattern in a normal ear; the normal ear provides the sort of pattern for which the theory was developed.

With the new aid, the subject could no longer use the former, distorted, memory-stored patterns for identification. He therefore had to learn a whole new set of patterns by association and a more or less trial and error process.

This then is the explanation offered: The first test came when the subject was still using his distorted patterns for a recognition criterion identification -- what this criterion might have been is not known. The second test shows some intermediate step in the learning process. Tests three and four came at the end of the learning process and show the results expected as based on theory.

This person has come to rely on the new aid for creating normal neural pattern shapes -- he would probably be temporarily upset if forced to re-employ former distorted memory-stored patterns.

It is well to note that, even with erroneous recognition, the diplacusis still charted the areas of nerve damage. Each curve implies that an area of maximum damage lies between three and four thousand cycles and that the damage falls off from this maximum in both directions.

The fourth test was the only one taken on subject 6 with the higher frequencies included and the curves show also that there is more loss at the higher frequencies.

Curve seven shows the average of the final two tests on subject six along with his threshold audiogram. As may be seen the diplacusis rather accurately maps the nerve damage as charted by the threshold audiogram except at the 5 kc point. It may be noted that at 5 kc the diplacusis audiogram implies no damage to the membrane while the threshold audiogram shows a high level of damage. The explanation is the same as that for curve five for much the same phenomenon.

4.4 Additional Comments on the Data

An attempt was made to relate the amount of diplacusis to db threshold loss. Unfortunately our sample was too small to produce any useable results since individual variations in the curves were large. Testing on a much larger scale is required in order to acquire accurate correlations.

CHAPTER FIVE

FUTURE DEVELOPMENT

It would be well here to make a few suggestions concerning future development of the audiometer and its testing.

Rather than having a continuously varying standard tone, in a "second generation" audiometer it might be better to have fixed tones that can be selected with a multi-position switch. It has been found that the frequencies tested give a relatively complete picture of the midrange of the cochlear damage; these should be included, perhaps along with several tones of higher frequency. The matching process becomes relatively more difficult at the higher frequencies but the inclusion of up to 10 and 12 kc tones in the audiometer should be seriously considered.

Although low frequency nerve deafness is rare, for a complete diplacusis audiogram, several of the lower frequency tones might be included. These should be in the range of 200 to 500 cps to the 1.5 kc suggested above. Thus in nine or ten fixed frequency steps, the entire basilar membrane may be checked for diplacusis.

In the constructed audiometer a manual switch was used by the subject to switch between the unmasked and partially masked tones.

This proved adequate except that the switch tended to make poor contact after long operation. Also there was enough capacitance in switching between switch positions so that, unless the switching was done fairly rapidly, an undesirable noise was heard.

To obviate these difficulties and also to allow the subject to concentrate his full attention on matching the two tones, the inclusion of an automatic switching procedure is suggested for any future device. This may be achieved by the use of an astable multivibrator with a suitable relay in the plate circuit. This would provide not only advantages of rapid switching but in addition the subject would not be bothered with having to devote part of his attention to the switching operation.

If the device is to be entirely self-contained, as it should be, there should be included within the audiometer an amplifier of sufficient gain and linearity to provide a signal sufficient to reach at least 25 db above threshold for any subject. The present unit has sufficient signal strength for most cases but the use of a separate amplifier is required in cases of extreme loss.

Another desirable feature might be the inclusion of separate attenuators in the signal and noise circuits. These would be calibrated in db so that after the threshold for each was found they could be combined in any manner for experimental purposes.

One other suggestion for improving the testing system is to use a more adequate pair of headphones. Several subjects commented that they could change the intensity of the sound by means of slight adjustment of the headphone on the ear. It is believed that this results from impedance mismatch between the ear cavity and the headphone. Possibly another type headset would eliminate this difficulty.

To further establish the validity of the theory, testing should be done on a clinical basis with subjects known to have nerve damage. In this way, a large body of data could be collected so as to provide adequate statistical measures.

Testing on a wide basis would also result in suggested improvements in the audiometer that would more readily adapt it to clinical use.

CHAPTER SIX

CONCLUSIONS

In this paper we have stated some basic principles and conclusions from the theory for hearing. These principles for the most part are generally acknowledged to be facts that have been substantiated by considerable experimental evidence. Starting from this basis we have then projected theory to the specific case of monaural diplacusis. We have tried in our theory to show how this diplacusis may be related to nerve damage in the cochlea. Also suggested in this development is a method of testing for diplacusis that may provide a reasonable check on the validity of the original hypothesis.

For experimental testing a monaural diplacusis audiometer was designed and built. This device has within it the means to provide both the desired pure tone, a pure tone corrupted with white noise, and a method for measuring the subjective pitch change as a result of the noise addition.

Results have been shown which are believed to support proposed theories concerning the manner in which a sound is recognized and the action of the cochlea. Empirical results agree well with theory and also are consistent with data taken from the electrical analog cochlea. Thus both the theory and the action of the analog in this situation have been confirmed by experimental data.

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