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THE PERCEPTION BY NORMAL HEARING PERSONS OF INTENSITY FLUCTUATIONS IN NARROW BAND STIMULI AND ITS IMPLICATIONS FOR SOUND FIELD CALIBRATION PROCEDURES

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ABSTRACT

Narrow band complex stimuli (such as F.M. tones, A.M. tones, damped wavetrains and narrow band noise) are used in sound field testing to overcome the effects of standing waves in the test enclosure. All such stimuli have a fluctuating intensity when they reach the ears of the person being tested. The temporal integration model proposed by Plomp and Bouman (1959) is suggested for use in the calibration of such fluctuating stimuli. Experimental data are given which show that when this is done, the threshold for pure tone stimuli may be inferred from measurements made with the complex stimuli. This equivalent pure tone threshold can be most accurately determined if the complex stimulus contains only small intensity fluctuations, or if the fluctuations occur at a rate sufficiently fast that the high intensity portions are not fully perceived by the ear. A method of calibration for such stimuli is suggested which involves reading the peak deflection of a standard sound level meter set to its "rms – fast" mode.

1. Introduction

A. The Problem

The determination of hearing thresholds in a sound field is complicated by the presence of room modes (standing waves) in the enclosure. These cause the acoustic gain curve (frequency response) from the speaker to the subject's ear to be a rapidly varying function of frequency (Morse, 1936). Furthermore, the actual frequency response depends heavily upon the position of both the loudspeaker and the subject. Errors in threshold determination can thus be caused by slight imprecision in either the stimulus frequency or the position of the subject during the measurement procedure or of the microphone during the calibration procedure. This has led to the suggestion that stimuli wider in bandwidth than a pure tone be used for sound field measurements (e.g. Morgan et al, 1979). Stimuli that have been suggested for use in the sound field include:

1. Frequency modulated (F.M.) tones (warble tones), (e.g. Reilly, 1958; Robinson and Vaughn, 1976).
2. Amplitude modulated (A.M.) tones, (e.g. Goldberg, 1979).
3. Damped wavetrains (DWTs), (e.g. Victoreen, 1974).
4. Narrow bands of noise, (e.g. Gengel et al, 1971; Barry and Resnick, 1978). Note that the waveform of a narrow band noise has the appearance of a continuous tone varying randomly in both frequency and amplitude.

The effects of these different stimuli on the distribution of intensity within the test enclosure are dealt with elsewhere (Dillon and Walker, in press). We have argued in that paper that the important stimulus parameter in relation to field homogeneity is signal bandwidth. This paper is addressed to the question of the perceptual effects of the different stimuli and the implications these have for calibration procedures.

After the above signals have passed through a transmission system of irregular frequency response (such as a loudspeaker and room), they have one feature in common: a variation of intensity with time. For the latter three stimuli this is easily understood since each stimulus contains temporal fluctuations before transmission through the test enclosure. For the F.M. tone, the intensity of the signal applied to the loudspeaker does not vary with time. However, as the tone sweeps in frequency, peaks and troughs in the room's frequency response are sequentially excited. The resultant signal received by the subject thus varies with time in both frequency and amplitude.

How does one deal with these fluctuations when calibrating the equipment and the room? Is it appropriate to choose the peak, rms, average, or some other function of either the intensity or the pressure to which the microphone is responding? In order that the clinical measurement may tell us something about the hearing of the person being tested and not about the test stimulus being used, we clearly want the calibration method to respond to the same properties of the stimulus as does the subject. Statements about the subject's sound field hearing threshold will not then have to be qualified by the type of stimulus or the type of averaging used. It will also be possible to state what the threshold would have been had it been possible to use a pure tone stimulus. To accomplish this we need to know how the temporal fluctuations are perceived. Some predictions may be made about the appropriate type of averaging by examining the known temporal integration properties of the ear.

B. Review of the literature on the perception of brief tone bursts

The perception of very brief tone and noise bursts has been extensively studied. The most consistent finding is that the loudness and/or threshold of an isolated tone burst changes by 10 dB for each decade of duration up to a value of about 200 ms. For durations in excess of 200 ms, threshold and loudness are independent of duration. These results have been obtained under a wide range of experimental conditions, although differences have been noted by some authors. In the measurement of loudness for example, the break-point (beyond which duration has no further effect) has been found to be smaller for more intense stimuli (Miller, 1948). In the measurement of threshold, the 10 dB per decade law has been found to apply down to durations as short as 1 ms (Garner, 1947a; Scholl, 1962), although some claim deviations from the law below about 10 ms (Garner, 1947b; Olsen and Carhart, 1966). The break-point in the threshold curve has also been shown to depend somewhat upon frequency, the time constant being smaller for the higher frequencies (Plomp, 1961), and upon the silent interval between signal bursts (Pollack, 1958). The change of threshold at a rate of 10 dB per decade of duration implies that, for brief stimuli, the threshold is determined by the total energy of the signal and is independent of its temporal distribution. Clearly the total energy of such stimuli must be determined by some sort of intensity integrator operating in the auditory system.

C. The leaky integrator model

Plomp and Bouman (1959) have shown that the experimental results can be well fitted by a "leaky integrator" model. It is as if a signal proportional to the intensity of the acoustic stimulus has been passed through a single time constant, low pass filter before being processed by the loudness or threshold determining sections of the auditory analyser. The time constant of the low pass filter is, of course, the 200 ms measured by so many of the previous investigators. The applicability of this model for isolated tone bursts has since been confirmed by several authors (e.g. Zwislocki, 1962; Zwicker and Wright, 1963). The model is shown in figure 1. A signal proportional to pressure is input at point A. The signal at B following the square law device is therefore proportional to intensity. Low pass filtering by the resistor and capacitor then produce the output signal at C. The input signal is assumed to be at threshold whenever the peak of the output signal just exceeds a certain criterion value. The criterion value is readily determined by applying a continuous tone of threshold intensity to the input and noting the (constant) value at the output.

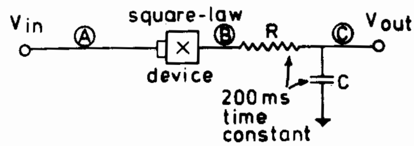

FIG. 1. Model of temporal summation for brief tones suggested by Plomp and Bouman (1959).

Although the model works well for isolated pulses, the agreement with empirical data is not so good for a large number of pulses with a high repetition rate (e.g. Zwislocki, 1962). In order to take account of additional data concerning the loudness of repeated tone bursts, Port (1963) developed a model similar to that of Plomp and Bouman's, but with a second, much larger discharge time constant. This has the effect of causing the excitation arising from a tone burst to build up more quickly than it dies away.

D. Application to sound field testing stimuli

As previously mentioned, the stimuli that have been proposed for sound field testing fluctuate with time (either cyclically or randomly). For very slow fluctuations, the peak values of the intensity will be maintained for a long time. If this time is sufficiently long, the peak intensity will pass through the low pass filter of figure 1 unaltered, and so the peaks of the stimulus will determined the intensity at threshold. For very fast fluctuations, the output of the filter will not change appreciably with time, and will have a value equal to the average intensity of the incoming signal. Thus the threshold will be determined by the average signal intensity. For intermediate fluctuation rates, the equivalent pure tone threshold intensity will have a value somewhere between the average and peak values of the stimulus.

Quantitatively, a very slow fluctuation rate is one in which the duration of the peak intensity is sufficiently in excess of the filter time constant. For a time constant of 200 msec this corresponds to a rate of sufficiently less than 5 Hz. A precise value is difficult to state since there may be several peaks and troughs in each stimulus modulation period. For the fluctuation rate to be considered to be very fast, the period must be sufficiently less than the filter time constant. This corresponds to a rate of well in excess of 5 Hz. Again a precise value cannot be stated since it depends on the variation of intensity that occurs within each stimulus modulation period.

Since stimuli are typically amplitude and frequency modulated at about a 5 Hz rate (Staab and Rintelmann, 1972), or pulsed on and off at a 2 Hz rate (Victoreen, 1974), or fluctuate randomly at up to about 5 Hz (for a 1/3 octave band centred at 1 kHz), it can be seen that neither of the two simplifying extremes outlined above can be applied to the stimuli that are normally used. A closer look at the threshold values of such stimuli is thus demanded.

2. Experimental Studies

A. *Overview of studies*

It is possible that the time constant (or constants if a two time constant model is appropriate) vary from subject to subject and from frequency to frequency. Thus, despite the extensive data that exists on the temporal summation of tone bursts, a new experiment is required to determine whether the threshold of an arbitrarily varying narrow band signal can be predicted for a particular subject on the basis of his continuous tone threshold, and time constant (or constants) alone. Additionally, since the apparatus shown in figure 1 is not available to investigators determining sound field thresholds in a clinical setting, we wished to determine whether any of the averaging or peak responding measuring techniques normally available would provide a satisfactory approximation during the calibration process.

The experiment involved three stages: 1) determination of the time constant(s) for each subject; 2) measurement of the thresholds for continuous tone, FM, DWTs, and narrow bands of noise for each subject; 3) comparison of predicted and measured thresholds.

B. *Individual time constant determination*

1. *The stimulus*

A suitable stimulus for determination of the time constants is shown in figure 2(a). This is a pure tone burst, the envelope of which is shown in figure 2(b). The output C of the model (presented in figure 1) is shown in figure 2(c). If the signal shown is at threshold, the value I_∞ corresponds to the intensity of a

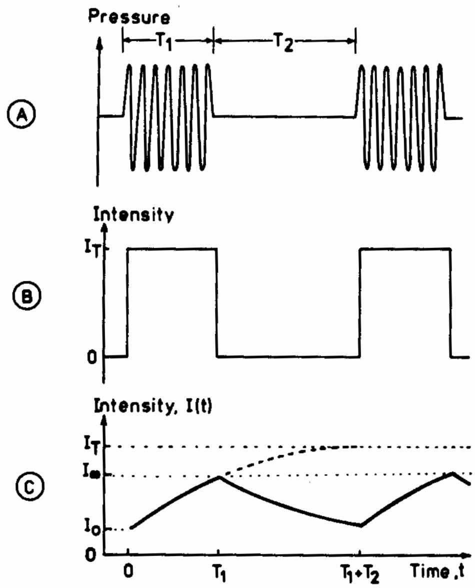

FIG. 2. The stimulus for time constant determination is shown in (a). Part (b) shows the envelope of the stimulus and part (c) the output when the stimulus is input to the network shown in figure 1.

continuous (or very long) tone also at threshold. This follows since the curve shown in figure 2(c) will have time to reach its asymptotic value for very large values of T_1 (i.e., $I_\infty = I_T$ where I_T is the intensity of the tone during the burst). Equations for the intensity, I(t), of the curve shown in figure 2(c) may be written as follows:

$$I(t) = I_o + (I_T - I_o) \cdot (1\text{-}\exp(-t/\tau_1)) \quad t < T_1 \quad \ldots.1$$

$$I(t) = I_\infty \cdot \exp\left(-\frac{t-T_1}{\tau_2}\right) \quad T_1 < t < T_1 + T_2 \quad \ldots.2$$

where τ_1 and τ_2 are the time constants corresponding to the rising and falling portions of the waveform respectively. By substituting the values of T_1 and $T_1 + T_2$ for t, we can solve the equations to show:

$$\frac{I_T}{I_\infty} = \frac{1\text{-}\exp(-T_2/\tau_2) \cdot \exp(-T_1/\tau_1)}{1\text{-}\exp(-T_1/\tau_1)} \quad \ldots\ldots\ldots.3$$

If T_2 is held constant to a value much greater than τ_2, equation 3 reduces to:

$$\frac{I_T}{I_\infty} = \frac{1}{1\text{-}\exp(-T_1/\tau_1)} \quad \ldots\ldots\ldots.4$$

Equation 4 may be rearranged to allow the signal intensity to be expressed in dB SPL as follows:

$$L_T = L_{\infty} - 10 \log (1\text{-}\exp(-T_1/\tau_1)) \quad ....5$$

where L_T is the SPL of the continuous tone from which the brief tone at threshold is gated, and L_{∞} is the SPL of a continuous tone at threshold.

2. *Experimental method*

Two series of experiments were performed to enable determination of the time constants. In the first, the stimulus OFF-time, T_2, was held to a large value (1 sec), and the stimulus ON-time, T_1, was varied over a wide range and the signal threshold found for each value. In the second series, T_1, was held to a short value (10 ms) and T_2 varied over a wide range. The equipment used to generate the stimuli and measure the hearing thresholds is shown in figure 3. Signals from the sine-wave oscillator (Hewlett Packard 3312A) were gated with a rise-fall time of 2.5 ms by an electronic switch (Grason-Stadler 1287). The subject maintained the signal at threshold by controlling a constant frequency Bekesy type recording attenuator (Grason-Stadler E3262A). Stimuli were delivered to the left ear through Telephonics TDH-49 supra-aural headphones. Testing was performed at the standard audiometric frequencies of 0.25, 0.5, 1, 2 and 4 kHz. Due to the large number of thresholds to be determined, the two authors (aged 29 and 37 years) served as the sole subjects. One subject had a threshold of 15 dB HTL at 4 kHz. Otherwise all conventional thresholds were within 10 dB of 0 dB HTL. Testing was performed in blocks. During each block, the ON and OFF times were held constant and the frequency was varied randomly.

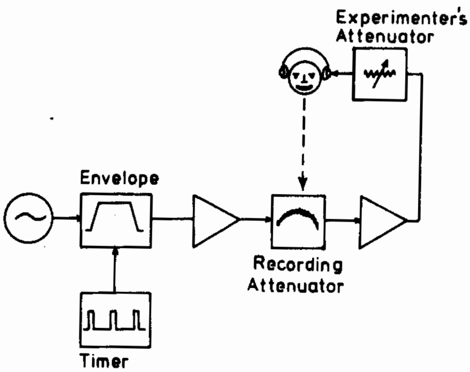

FIG. 3. Equipment arrangement for time constant determination and pure tone testing.

3. *Results*

a) *Variable ON-time*

A typical set of thresholds for one subject at one frequency is shown in figure 4. The smooth curve is a minimum least squares fit to the data based on equation 5. Inspection of equation 5 shows that two parameters need to be estimated before such a curve can be drawn. These are the threshold SPL of a continuous (or sufficiently long) tone and the time constant, τ_1. Estimation of these parameters was carried out by means of an iterative least squares procedure. The longer duration stimuli were measured several times as a small error in the estimation of the continuous tone threshold can have a large effect on the time constant determination.

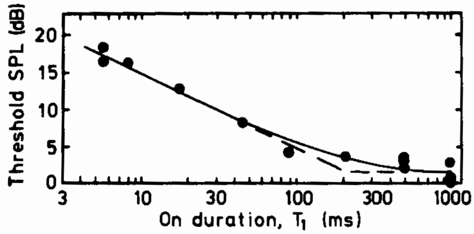

FIG. 4. Typical set of thresholds for different ON durations (for subject H.D. at 1 kHz). The solid curve shows the least squares fit to the data, and the broken line shows how the time constant is equal to the break point in the curve.

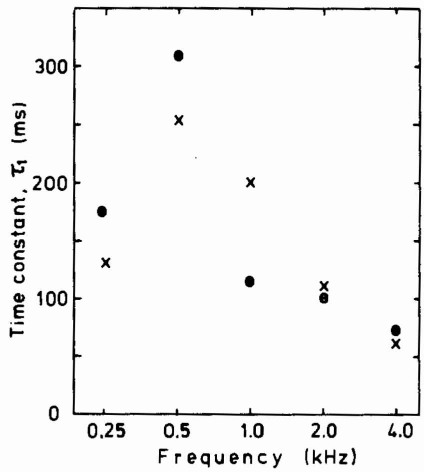

FIG. 5. Values of the time constant, τ_1, obtained from the first experimental series. Symbols: • – G.W.; X – H.D.

The results of ten such curves are shown in figure 5. It is apparent that the time constant, τ_1, is smaller at the higher frequencies. Figure 5 also indicates that for both subjects, the time constant at 250 Hz is lower than at 500 Hz. However, it should be noted that unless each point on the curve of threshold versus duration (e.g. figure 4) is known with some precision, the estimation of the break point is quite inexact. An error of 2 dB in the estimation of the continuous tone threshold, for example, will lead to a 50% error in the time constant determination! Fortunately, this does not make the measurements meaningless since the converse must also be true: accurate estimation of the time constant is not required if the threshold of a brief signal is to be predicted to within acceptable accuracy. From figure 5 we conclude that the time constant appears to decreased with increasing stimulus frequency, and that for the subjects measured, the much quoted value of 200 ms is an acceptable approximation for the three lower frequencies, but is probably too large for the two higher frequencies.

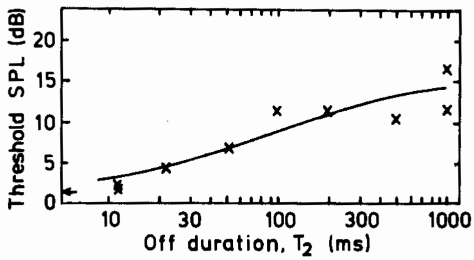

FIG. 6. Typical set of thresholds for different OFF durations (for subject H.D. at 1 kHz). The solid curve shows the least squares fit to the data. The arrow shows the continuous tone threshold.

b) Variable OFF-time

These conclusions are supported by the second series of experiments. The stimulus ON-time, T_1, was held constant at 10 ms and the OFF-time, T_2, varied between 10 ms and 1 sec. A typical set of experimental points and a fitted theoretical curve (from equation 3) are shown in figure 6. The curve flattens out to a plateau at each extreme of T_2. For small values of T_2, the stimulus approaches a continuous tone, while for large values there is no interaction between successive tone bursts, and further increases in T_2 have no effect. Three parameters are necessary to determine the theoretical curve, L_∞ and the two time constants, τ_1 and τ_2. For each subject, the value for L_∞ (the continuous pure tone threshold) was set to that found in the previous series, while τ_1 and τ_2 were varied iteratively to provide a least squares fit to the data.

The resulting values are shown as a function of frequency in figure 7. A decrease of τ_1 with increasing frequency is again apparent from this independent set of results. Values for the second time constant, τ_2, show no systematic trend. Estimation of τ_2 is an even more inexact process than it is for τ_1. Small changes in either τ_1 or L_∞ can produce quite large changes in the value estimated for τ_2. Once again, however, the converse of this means that accurate values of τ_2 are not required. Significant errors (caused by uncertainty of the correct value of τ_2) in the

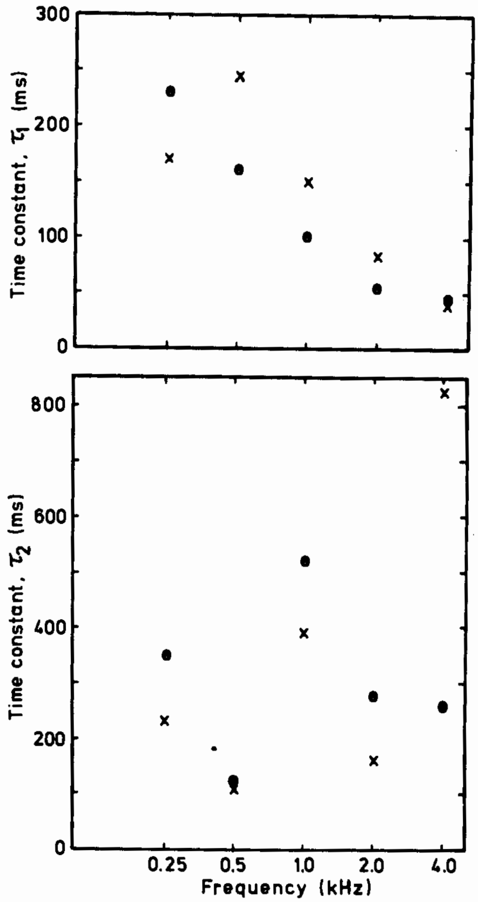

FIG. 7. Values for both time constants as a function of frequency. Symbols: ● – G.W.; X – H.D.

prediction of the threshold of brief or fluctuating stimuli will occur only when the stimulus intensity peaks have a duration less than the value of τ_1, and when the stimulus has a small intensity for a time comparable to τ_2. Because of these considerations it was decided that a single time constant (equal to τ_1) would be used for all subsequent work. Clearly, for these two subjects at least, the value of this time constant will need to vary with frequency.

Despite differences in methodology and the inexactness of the measures, the present data on thresholds agrees surprisingly well with that of Port (1963) on loudness. Port's values for τ_1 and τ_2 for a 2.5 kHz tone were 70 and 200 msecs respectively compared to our average at 2 kHz of 87 and 217 msecs.

C. Measurement of thresholds

The second part of this study involved the measurement of the thresholds for continuous tones, FM tones, narrow band noise, and damped wavetrains (DWT's) for each subject. AM tones were not included as these had already been rejected as unsuitable for sound field testing on other grounds (Dillon and Walker, in press).

1. Method

The experimental set-up was as previously described with the following additions. The DWT's were generated by passing a 1 Hz square wave through the band pass filter of a frequency analyser (B&K model 2121). The FM tones had to be generated in such a way that the amplitude modulation which occurs when the tone is used in the sound field is included. This was achieved by delivering the FM signal to a loudspeaker in the audiometric test enclosure and tape recording the output of the microphone of a Kemar manikin (Burkhard and Sachs, 1975) also located in the room. The tape recorded signal was then delivered through the earphones to establish the thresholds. It is appreciated that the FM signal is not completely natural in that it will have passed through the ear canal twice (i.e., the ear canal of Kemar and the ear canal of the subject). This will not affect the results since our aim was to obtain a signal simultaneously modulated in both frequency and amplitude.

The actual stimuli used were: (a) Pure tones with frequencies of 0.25, 0.5, 1, 2 and 4 kHz; (b) FM tones with a modulation rate of 6.3 Hz and the deviation varied to provide a constant bandwidth of ±5% at the same centre frequencies as in (a); (c) FM tones with a centre frequency of 1.5 kHz and deviations of 16, 40, and 100 Hz at each of the modulation rates, 2.5 and 10 Hz. This particular centre frequency was used because previous work indicated that the room would introduce reasonably large amounts of amplitude modulation around this frequency. It thus provided a critical test of our hypothesis; (d) DWT's with 3% bandwidth and the same centre frequencies as in (a). This bandwidth is the one suggested by Victoreen (1974); (e) Narrow band noise with a centre frequency of 1 kHz and bandwidths of 1, 3, 10, 23, 29 and 70%.

The threshold values obtained for the pure tones, expressed as the SPL generated in a 6cc coupler (B&K model 4152) are shown in figure 8. The differences between these thresholds and those for the other stimuli are reported in the following section where they are compared to the theoretically predicted differences.

D. Comparison of predicted and measured thresholds

1. Rationale

In order to compare predicted and measured thresholds, each of the stimuli was input to the temporal integration model previously described, with the time constant of the model set to match that of each subject at each centre frequency (using data obtained in the first stage of this experiment). For each input, the model will give a slowly varying waveform at its output. If the model is valid, the peak values of the outputs for all of the complex signals at threshold should have the same value as that obtained for a pure tone, also at threshold. Another way of expressing this is that the difference between the peak to peak levels of the sine wave and the complex wave when both give the same peak output from the model should be equal to the difference between the peak to peak level of the two signals when both are at threshold. This prediction has been tested for the FM and DWT stimuli.

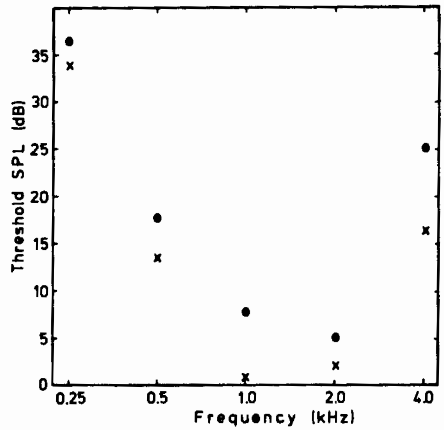

FIG. 8. Pure tone thresholds for both subjects. Obtained for pulsed tones with T_1 = 1 sec, T_2 = 1 sec. Symbols: • – G.W.; X – H.D.

2. Results

a) Comparison of measured and predicted results for DWTs.

Figure 9 shows, for each subject, the predicted (model) and experimentally determined differences in peak to peak level between the DWT and pure tone stimuli when both are at threshold. Fairly large differences in peak to peak level (of the order of 10 dB) are both expected and obtained since the DWT's consist of quite brief pulses, especially for the higher frequencies. As frequency increases, the duration of each DWT pulse decreases more rapidly than does the measured time constant, thus explaining the rising nature of the curve obtained from the model. A rising curve with a similar order of magnitude is also obtained with subjectively measured thresholds, although there are clearly some differences between the model results and the subjective results.

To assess the significance of these differences, we need to estimate the uncertainties associated with each curve. Whenever repeated measurements were made in these experiments, we found a standard

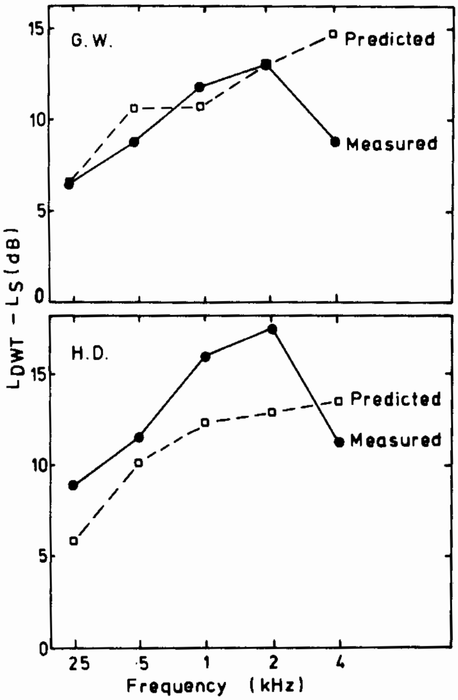

FIG. 9. The difference between the peak to peak level of the DWT stimulus, L_{DWT}, and the peak to peak level of the pure tone, L_S, when both are at threshold ("measured") or produce the same peak output from the model ("predicted").

deviation of around 2 dB. This is smaller than the 3 dB obtained by Erlandsson et al (1979), which is probably due to our use of experienced subjects. Since the experimental curve in figure 9 represents the difference between two thresholds (each respresenting the average of two measurements) we can expect the one standard deviation limits of the points shown in figure 9 to be the same as that for a single measurement (averaging two measurements reduces the SD by a factor of $\sqrt{2}$ but taking the difference between measures increases it by the same amount). Hence the standard deviation of the threshold difference is 2 dB. Discrepancies of 4 dB can thus be expected occasionally due to this source of error alone. The model curve is exact to within ±0.2 dB provided no error has been made in the time constant determination. However, this is by no means the case, and we estimate a possible error of up to 2 dB for the points on the model curve. Thus, the model and subjective results shown in figure 9 agree to within experimental error for subject GW. For subject HD, although the results are sufficiently close when each frequency is examined independently, the consistent elevation of the subjective curve for four of the five frequencies may indicate a small systematic deviation from the model. Since the DWT stimuli are very brief and contain large amplitude fluctuations within each cycle, the prediction of their thresholds from a knowledge of the pure tone thresholds is a critical test of the model. We conclude that the model is successful in estimating the thresholds for DWT's to within about 5 dB.

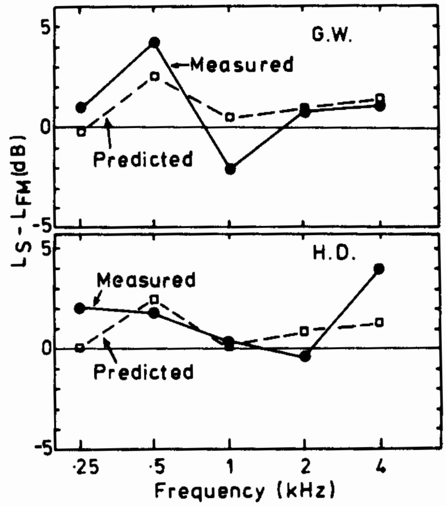

FIG. 10. The difference between the peak to peak level of the pure tone, L_S and the peak to peak level of the FM tone, L_{FM} when both are at threshold ("measured") or produce the same peak output from the model ("predicted"). A frequency deviation of ±5% and a modulation rate of 6.3 Hz was used. The levels are measured electrically at the point of stimulus generation.

b) Comparison of measured and predicted results for F.M. tones.

Results for FM stimuli are presented in figure 10. Agreement between the model and subjective results is good, (the same experimental uncertainties as before are operating). Results for FM stimuli of different deviations and modulation rates are shown in figure 11. Both the model and subjective results indicate that FM signals with a large frequency deviation require lower level signals to be input to the loudspeaker than for pure tones if both signals are to be at threshold. Further, the effects can be seen to be larger at low modulation rates than at high ones. Both of these effects have been shown many times before (e.g. Young and Harbert, 1970; Barry and Resnick, 1978), but with the insight provided by the model we can suggest several reasons for this behaviour.

We have previously outlined the mechanism by which FM tones acquire an amplitude modulation due to loudspeaker response and room transmission effects. A typical frequency response (at a point 1.2m from the speaker)

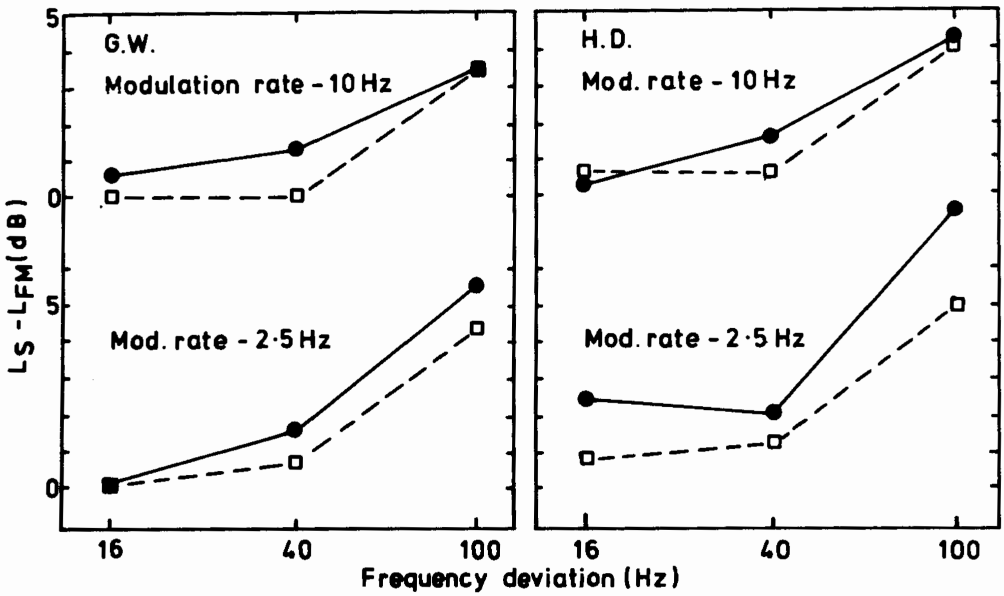

FIG. 11. The same as figure 10, only for a range of frequency deviations and modulation rates. A centre frequency of 1150 Hz was used. Symbols: Measured ●—●; predicted from model □- - -□

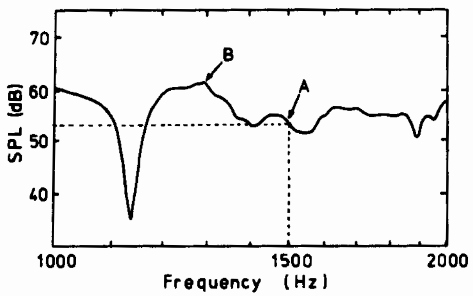

FIG. 12. A typical frequency response at a fixed point in the sound field for the frequency range 1 to 2 kHz.

is shown in figure 12. For a pure tone at 1.5 kHz, a SPL of 53 dB will be measured at this point ("A" in figure 12). When the frequency of the tone is modulated by a small amount, lower frequencies will produce higher SPL's and higher frequencies lower SPL's (in this instance). As the frequency deviation is made larger, there is an increasing likelihood that the swing will extend over a region of the response that produces a higher SPL than does the original centre frequency. If a low modulation rate is used, the frequency will linger in these more responsive regions for a time sufficiently long that:

(i) The room modes in this region will be excited. (A time of the order of 50 ms is required for the test booth used in this experiment.
(ii) The subject will be able to fully utilise this high intensity portion of the modulation cycle. (As we have seen, a time constant of 200 ms or less is appropriate for threshold detection.)

Under these conditions, the high intensity part of the modulation cycle will largely determine the threshold for the whole stimulus. This lowering of the threshold (compared to a pure tone at the centre frequency of the complex) is thus most likely to occur for large deviations and slow modulation rates. Notice that the objective in using FM tones (the avoidance of room transmission effects) has not been realised under these conditions since

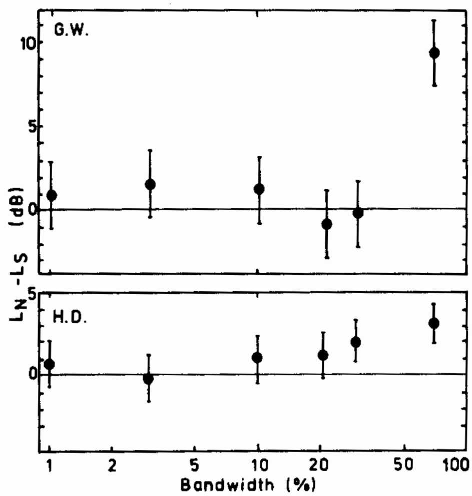

FIG. 13. Difference in rms levels between narrow band noise of various bandwidths and a pure tone when both are at threshold. A centre frequency of 1 kHz was used.

individual peaks have the same effect as they do for pure tones.

If our hypothetical threshold measurement has been made at a frequency of 1280 Hz (point "B" in figure 12), the threshold measured for the FM stimulus could be expected to be *higher* than that for the pure tone. As a frequency of 1280 Hz results in the highest SPL between 1 and 2 kHz, the spread of energy is into frequency regions of lower sensitivity (for this particular room transmission response). Such a case is clearly the exception rather than the rule. Thus based on physical factors alone we would expect thresholds for FM stimuli to usually be lower than those for pure tones.

The effects of the modulation parameters discussed above also apply to testing under earphones as it has been shown that when a constant voltage signal is applied to the input of audiometric earphones the frequency response measured at the earcanal entrance is not flat for frequencies below 0.5 kHz or above 2 kHz (Shaw, 1966). For low and high frequencies, this frequency dependent response will then lead to the same effects as does the loudspeaker response and room modes in sound field testing.

c) Noise – a special case

Due to its random nature, narrow band noise proved to be of little use in evaluating the model being discussed in this paper. Predicted values for the threshold of each complex stimulus are obtained by observing the peak value at the output of the threshold model. Unfortunately, noise has no single peak value and any prediction would necessarily be based on an arbitrary choice of peak value. Consequently, only the experimentally determined differences are shown in figure 13.

The measurements were performed with a centre frequency of 1 kHz. The levels for both the pure tone and the noise were measured on an rms responding meter set to its slow response mode (B&K model 2107). Thresholds for each type of stimulus were measured three times and the means determined. The error bars shown in the figure correspond to plus and minus twice the standard error of the mean difference between the narrow band noise and pure tone stimuli. No difference in the thresholds of the two stimuli are observed for bandwidths up to about 20%. This is in accord with Gassler's (1954) observation that the total energy within a critical band contributes to the threshold of a narrow band signal. For normal hearing persons the critical band is usually taken to have a value of 16% at a centre frequency of 1 kHz (Scharf, 1970).

Wit (1978) carried out a exercise similar to ours with damped wavetrain signals and a different temporal integration model (the diverted input model – Garner and Miller, 1947). His measured and predicted results are also in close agreement, particularly when a time constant of 85 msecs was applied to the model. However, the procedure used by Wit cannot easily be applied to continuous stimuli such as FM tones.

Our concept of different signals all being at threshold when they produce the same peak output from the model is indirectly supported by the results of Yost and Klein (1979). Three types of narrow band stimuli were found to have the same total energy when each was at threshold. Since each stimulus was considerably shorter than the 200 ms time constant of the model, all of the energy in each stimuli would also be integrated by the model. This implies that the model would produce the same peak output for all three stimuli.

3. A Suggested Calibration Procedure for use in the Audiological Clinic.

Our model is reasonably successful in predicting the threshold for time varying narrow band stimuli from the threshold of a pure tone of the same frequency. Conversely, if the network shown in figure 1 is used for the calibration of such stimuli in the sound field, the threshold obtained may be expressed as the SPL of a pure tone of threshold intensity. Comparison with other audiometric data is thus facilitated.

Although the model is useful in the laboratory as a calibration tool, it is clearly too complicated for clinical use. Firstly, for the most accurate use, the time constant needs to be determined for each subject. Secondly, the room has to be calibrated with these time constants incorporated in the calibrating equipment, necessitating a separate room calibration for each subject. Thirdly, the equipment shown in figure 1 is required.

Fortunately, the results of this paper can be applied to the calibration of sound field stimuli in such a way that the above difficulties are avoided. We have noted that reasonably large errors in the time constant will result in fairly small errors in the inferred equivalent pure tone threshold. Specifically, an error of a factor of two in the time constant will lead to a maximum possible error of 3 dB in the inferred threshold. This worst case error will occur for a stimulus which contains very large, and slow amplitude fluctuations. Stimuli which possess these characteristics include DWT's, narrow bandwidth AM tones, very narrow bands of noise (especially for lower centre frequencies), and FM tones with a large frequency deviation and a low modulation rate. If these types of stimuli are avoided, only small errors will be incurred, even if the time constant in the calibration equipment is markedly different from that appropriate to the person being tested. Thus it is only necessary that the room and equipment be calibrated once.

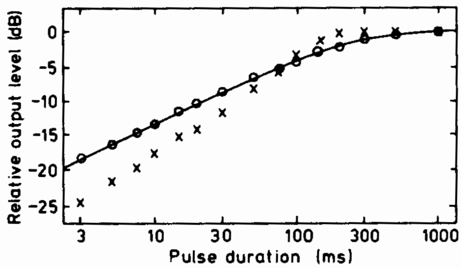

FIG. 14. Response of a B&K sound level meter (model 2107) to tone bursts of various durations (crosses). The desired response (theoretically calculated) is shown as the solid curve, and the measured response of the network shown in figure 1 as the circles.

The need for special equipment is also readily overcome. The equipment shown in figure 1 consists of a square law device, a low pass filter, and a device to enable observation of the output. These same components are present in any true rms responding sound level meter. Not all instruments marked "rms" do, in fact, employ a technique which enables the true rms value of a waveform to be measured (Wahrmann and Broch, 1975). The meters in the Bruel and Kjaer range of instruments, for example, use an approximation which provides satisfactory results for waveforms with crest factors less than 5 (i.e. reasonably non-peaky waveforms). Tone bursts of various durations were input to such an instrument (B&K model 2107) and the peak deflection of the meter noted for each duration. These results are shown as crosses in figure 14. A break point in the curve occurs at about 200ms. By comparison, the peak output that should result from a true rms device with a simple (1 st. order) low pass filter performing the averaging, is shown as the solid curve in figure 14. This curve is based on equation 5. The circles lying near the curve show the measured peak output from the network of figure 1 especially constructed for this experiment. These results show that while the "rms responding" meter does not exactly meet the present requirement, it does provide a reasonable approximation. The error in the inferred equivalent pure tone threshold will be largest for the same type of stimuli as discussed before, i.e. those with large and slow amplitude fluctuations. Stimuli which produce only a small wobble in the needle of the calibrating meter (set to "rms — fast") are thus recommended for sound field procedures if one wishes to accurately infer an equivalent pure tone threshold intensity. Certain constraints about the bandwidth of such stimuli must also be satisfied, but these are the subject of a second paper (Dillon and Walker, in press). Essentially, we have recommended the use of sinusoidally modulated FM tones with a modulation rate of 20 Hz and a frequency deviation chosen for each centre frequency to provide a desirable stimulus bandwidth.

In summary, we have argued in this paper that if a stimulus as described above is used in sound field testing, calibration of the room using the peak deflection of a standard sound level meter set to its "RMS — Fast" mode will permit the obtained thresholds to be expressed in terms of equivalent pure tone thresholds. A final word of caution is necessary. This data was collected with normal hearing subjects and it is clear that it needs to be validated for subjects with various patterns of hearing loss.

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