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, chirped waveforms and narrow band noise) are used in sound field testing to overcome the effects of rendering waves in the test enclosure. All such stimuli have a number of features in common: the stimulus has a limited bandwidth, it is a slowly varying function of frequency when plotted in a log frequency scale, and it is based on a general and simple theoretical model (the minimum (lognormal) model proposed by Friberg and Blount, 1969) that is suggested for use in the calibration of fluctuating stimuli. Important measures are given which show how when the stimulus is used, the processes for the F.M. and A.M. noise stimuli are related to the processes for the complex stimuli. The equivalent pure tone threshold can be most accurately determined in the narrow band stimulus and only small intensity fluctuations, or if the fluctuations occur at a sufficiently fast rate that the high frequency portions are not fully perceived by the ear. A method of calibration for both stimuli is suggested which involves needing the peak definition of a standard sound level meter set to its "slow - fast" mode.

1. Introduction

4. The Problem

The determination of hearing thresholds is 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 (Moore, 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 (marble tones), (e.g. Revill, 1968; Robinson and Vaughn, 1976);
2. Amplitude modulated (A.M.) tones, (e.g. Goldberg, 1978);
3. Chirped waveforms (CHW), (e.g. Victorian, 1974);
4. Narrow bands of noise, (e.g. Gargie et al, 1971; Barry and Reesnick, 1972). 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 varies 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

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by the loudness or threshold determining sections of the auditory analyzer. 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., Zwicker, 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 integrator 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 time or threshold intensity to the input and noting the (constant) value at the output.

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., Zwicker, 1962). In order to take account of additional data on the loudness of the burst, Port (1963) developed a model similar to that of Ploog 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 determine the intensity at threshold. For very fast fluctuations, the output of the filter will not change appreciably with time, and we 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.
Quantitative; a very slow fluctuation rate is one in which the duration of the weak intensity is sufficiently in excess of the filter time constant. For a time constant of 200 ms, 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 Tentschert, 1932), or pulsed on and off at a 2 Hz rate (Vicuñen, 1974), or fluctuate randomly at up to about 5 Hz (for a 1/3 octave band centered 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 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 arbitrary varying narrow band signal can be predicted for a particular subject on the basis of his continued tone, threshold, and time constant (or constants) above. 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 aspects: 1) determination of the time constant(s) for each subject; 2) measurement of the thresholds for continuous tone, FM, DMW, and sweep tone bands of noise for each subject; 3) comparison of predicted and measured thresholds.

B. Individual time constant determination

The stimulus

A suitable stimulus for determination of the time constant 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 of I corresponds to the intensity of a continuous (or very large) tone 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, i.e., I = I th where I th 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:

\[ I(t) = I_0 + I_1 - I_2 \times (1 - \exp(-T_1/T)) \]

where \( T_1 \) and \( T_2 \) are the time constants corresponding to the rising and falling portions of the waveform respectively. By substituting the values of \( T_1 \) and \( T_2 \) for \( T \), we can solve the equations to show:

\[ I(t) = I_0 \times (1 - \exp(-T_1/T_1)) \times \exp(-T_2/T_2) \]

If \( T_1 \) is held constant to a value much greater than \( T_2 \), equation 3 reduces to:

\[ I(t) = I_0 \times 1 - \exp(-T_1/T) \]

Equation 4 may be rearranged to allow the signal intensity to be expressed in dB SPL as follows:
where \( L_T \) is the SPL of the continuous tone from which the brief tone at threshold is gated, and \( L_{max} \) 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, \( T_{off} \), was held in a large value (1 sec) and the stimulus ON-time, \( T_{on} \), was varied over a wide range and the signal threshold found for each value. In the second series, \( T_{on} \), was held to a short value (10 ms) and \( T_{off} \) varied over a wide range. The equipment used to generate the stimulus and measure the hearing threshold is shown in figure 3. Signals from the sine-wave oscillator (Hewlett Packard 3312A) were gated with a rise-fall time of 2.9 ms by an electronic switch (Graun Stadler 1287). The subject maintained the signal at threshold by controlling a constant frequency 500-Ohm type recording attenuator (Graun Stadler ER222A). Stimuli were delivered to the left ear through Telephonics TDH-40 supra-aural headphones. Testing was performed at the standard audiometric frequencies of 0.26, 0.5, 1, 2 and 4 kHz. Due to the large number of thresholds to be determined, the two authors (aged 28 and 37 years) tested as the sole subjects. One subject had a threshold of 40 dB nHL at 4 kHz. Otherwise, all conventional thresholds were within 10 dB of 0 dB nHL. 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.](image)

3. Results

a) Variable ON-time

A typ e 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
the measurements meaningless since the converse must not 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
remain constant with increasing stimulus frequency and that for the subjects measured, the much
quoted value of 200 ms is an acceptable approximation for the
lower frequencies, but is probably too large for the two higher
frequencies.

[Graph showing threshold vs off duration]

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 error
bars show the continuous tone threshold.

b Variable off-time

These conclusions are supported by the
second series of experiments. The stimulus
on-time, T1, was held constant at 10 ms and
the off-time, T2, 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 T2. For small
values of T2, the stimulus approaches a
continuous wave while for large values there is
no interaction between successive tone bursts,
and further increases in T2 have no effect.

Three parameters were necessary for the data,
the theoretical curve, T1,0 and the two time
constants, τ1 and τ2. For each subject, the value
for T1,0 (the continuous pure tone threshold)
was set at the point found in the previous series,
while τ1 and τ2 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 τ1, with
increasing frequency is again apparent from
this independent set of results. Values for the
second time constant, τ2, show no
systematic trend. Estimation of τ2 is an even
more inexact process than it is for τ1. Small
changes in either τ1 or T1,0 can produce quite
large changes in the value estimated for τ2.

Once again, however, the converse of this
means that accurate values of T2 are not
required. Significant errors (caused by
uncertainty of the correct value of τ2) in the

[Graph showing time constant vs frequency]

FIG. 7. Values for both time constants as a function
of frequency. Symbols + - - O, X - - .

prediction of the threshold of brief or
fluctuating stimuli will only occur when the
stimulus intensity peaks have a duration less
than the value of τ1, and when the stimulus
has a small intensity for a tone comparable to
T2. Because of these considerations it was
decided that a single time constant (equal to
τ1) 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 threshold agree surprisingly well with that
of Port (1963) on loudness. Port's values for
τ1 and τ2 for a 2.5 kHz tone were 70 and 200
msec respectively compared to our average at
2 kHz of 87 and 217 msec.

C Measurement of stopbands

The second part of this study involved
the measurement of the thresholds for
continuous tones, FM tones, narrow band noise,
and damped wavelets (DWT's) for each subject.
All tones were not included as these had
already been rejected as unsuitable for sound field testing on other grounds (Dillons 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 kHz 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 (Rutland and Sear, 1975) also located in the room. The tape-recorded signal was then delivered through the earphone 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 0.3 Hz and the deviation varied to provide a constant bandwidth of 1.5% 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.

In the following the critical test of our hypothesis (d) DWT’s with 3% bandwidth and the three centre frequencies as in (a). This bandwidth is the one suggested by Wever (1974); (e) Narrow band noise with a centre frequency of 1 kHz and band widths of 1, 2, 10, 20, 30, and 70%. The threshold values obtained for the pure tones, expressed as the SPL generated in a 6 cc 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 data are compared to the theoretically predicted differences.

2. Comparison of predicted and measured thresholds

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 would 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 differences between the peak to peak levels of the DWT and the complex wave when both the same peak output from the model would 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.

2. Results

a) Comparison of measured and predicted results for DWT's.

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 uncertainty associated with each curve. Whenever repeated measurements were made in these experiments, we found a standard
deivation of around 2 dB. This is smaller than the 3 dB obtained by Erlendsson 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 representing the average of two measurements) we can expect the standard deviation limits of the two points shown, in figure 3 to be the same as that for a single measurement (averaging two measurements reduces the SD by a factor of \(\sqrt{2}\) but takes 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 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 positive error of up to 5 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 constant 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 threshold for DWT's ± within about 5 dB.

### Figure 10

The difference between the peaks to peaks level of the pure tone, i.e., where both are at threshold ("measured") or produce the same output from the model ("predicted").

#### 5.1 Calibration 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 levels 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. Most of these effects have been shown many times before (e.g., Young and Hartleit, 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 1100 Hz was used. Symboles: Measured (O); 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. (A) shows the SPL of 53 dB at this point. When the frequency of the tone is modulated by a small amount, lower frequencies produce higher SPL's, and higher frequencies lower SPL's. 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 utilize this high intensity portion of the modulation cycle. (As we have seen, a time constant of 200 ms or less is appropriate for threshold determination.)

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 low 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 level between narrow-band noise at 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 1260 Hz (peak "A" 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 1260 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 modulated stimulus, response). Such a case 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 described above also apply to testing under earphones if it is known that a constant waveform signal is applied to the input of audiometric earphones the frequency response measured at the earphone entrance is not flat for frequencies below 0.5 kHz or above 2 MHz (Shaw, 1986). For low and high frequencies, this frequency dependent responsivity will then lead to the same effects as those 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 described in this paper. Predicted values for the threshold of each complex stimulus are obtained by observing the peak values for 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 carrier frequency at 1 kHz. The levels for both the pure tone and the noise were measured on an oscilloscope using the "slow response mode" (IBM model 2107). Thresholds for each type of stimulus were measured three times and the mean 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 kHz. This is in accord with Gauger's (1964) 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 10% at a center frequency of 1 kHz (Scharf, 1976).

Wit (1979) carried out an exercise similar to our with damped wavelet signals and a different temporal integration model (the delayed input model — Garner and Miller, 1947). His measured and predicted results are also in close agreement, particularly when the time constant of 85 ms 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 signs 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 integral energy when each was at threshold. Since each stimulus was considerably shorter than the 20 ms time constant of the model, all of the energy in each stimulus 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 tone varying narrow band stimuli from the threshold of a pure tone of the same frequency. Conversely, if the waveform 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 at 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, which in the normal room is likely to be calibrated with the time constants incorporated in the calibrating equipment. Consequently, a separate room calibration for each subject, thereby, the model shown in Figure 1 is required.

Fortunately, the results of this paper can be used for the calibration of sound field stimuli in such a way that the above difficulties are avoided. We have noted that reasonable large errors in the time constant will result in fairly small errors in the inferred equivalent pure tones 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 world case error will occur for a stimulus which contains very large, and slow amplitude fluctuations, stimuli which possess these characteristics include DDKT, narrow bandwidth AM tones, very narrow bands of noise (especially for lower centre frequencies), and FM tones with a large frequency increment and a low modulation rate. If such stimuli are avoided, only small errors will be incurred, even if the time constant in the
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 components are seen present in any two 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 (1st 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 stimulus as discussed before, i.e., those with large and slow amplitude fluctuations. Stimuli with these characteristics would be a small variable in the sense of the calibrating meter (set to “rms - fan”), 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 20Hz 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 loudness 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.

References


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