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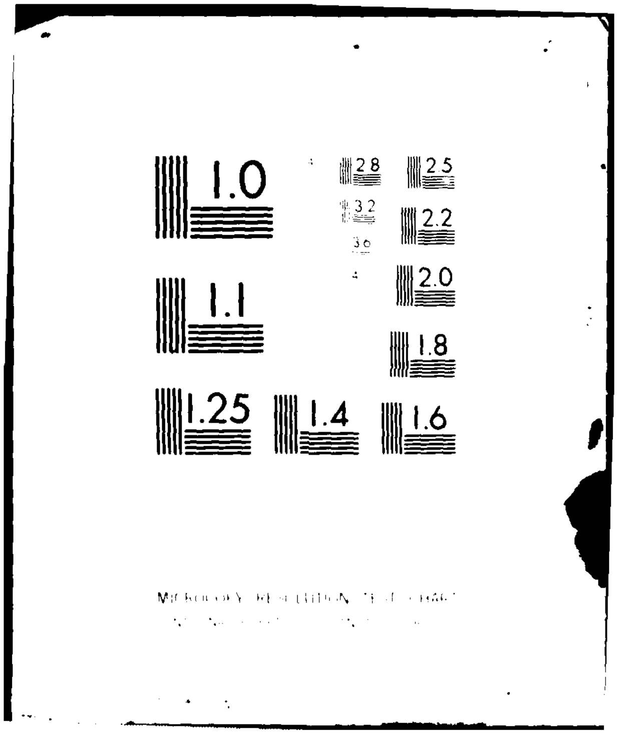
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TEMPORAL INTEGRATION IN LOW FREQUENCY AUDITORY DETECTION

Richard Horonjeff, Sanford Fidell,
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BOLT, BERANEK, AND NEWMAN INC.
Cambridge, Mass. 02138

March 1981

Final Report

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APPLIED TECHNOLOGY LABORATORY POSITION STATEMENT

This report provides insight into helicopter aural detectability problems. When approaching the threshold of detectability, a listening observer depends on the effects of both frequency and signal duration to enable him to distinguish the rotor system noise signature from the background ambient noise. The results of this study provide an understanding of these effects that will be incorporated into other R&D efforts at the Applied Technology Laboratory to improve aural detection prediction of Army aircraft.

Bill W. Scruggs, Jr., of the Aeronautical Systems Division served as project engineer for this effort.

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This report describes research undertaken to determine whether the Applied Technology Laboratory's existing procedures for predicting detection ranges of helicopters require revision to account for any potential differences in the time period over which people integrate signal energy at different frequencies. Findings disclosed that over the frequency range of 40 to 1000 Hz the integration time of the human auditory system was essentially invariant

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at approximately 240 milliseconds. Although no change to the ATL detection range program was required, incorporation of temporal effects was recommended to improve the generality of the program.

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INTRODUCTION

The noise signature of a distant helicopter when first aurally detected by an unaided human observer in the field is usually not much greater in level than the background noise in the observer's vicinity. As small fluctuations in signal levels (due to the vagaries of long range propagation of the helicopter's acoustic emissions through an inhomogeneous atmosphere) and/or fluctuations in ambient noise levels (due to random variations in the background noise process) perturb the signal to noise ratio at the observer's ear, the helicopter is discontinuously audible.

For example, when atmospheric conditions favor propagation of acoustic energy in the frequency range in which a helicopter is maximally detectable for a few hundred milliseconds, and when background noise levels happen to be simultaneously low, a distant helicopter may be momentarily audible. Moments later, however, a moving helicopter may become inaudible and remain so for several seconds for any of a large number of reasons. These include changes in source levels during various flight maneuvers (Reference 1), shielding from terrain features, random wind gusts, etc.

Algorithms for predicting ranges at which helicopters first become audible should therefore recognize the intermittent and short duration of audibility. This in turn requires consideration of the effects of duration of human audibility of

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[1]Lowson, M. V., and Ollerhead, J. B., STUDIES OF HELICOPTER ROTOR NOISE, USAAVLABS TR 68-60, U.S. Army Aviation Materiel Laboratories, Fort Eustis, Virginia, January 1969, AD 684394.

acoustic signals, and of the interactions between duration and signal frequency.

This report describes research undertaken to determine whether the Applied Technology Laboratory's existing procedures for predicting detection ranges of helicopters require revision to account for any potential differences in the time period over which people integrate signal energy at different frequencies.

LITERATURE REVIEW

The most common framework for considering effects of duration on the ear's sensitivity to acoustic signals rests on the assumption that the ear integrates the power of an acoustic signal over a fixed period of time. The period of integration is usually referred to as the time constant of the ear. Zwislocki's mathematical model of "temporal auditory summation" (Reference 2) provides a useful framework for the following discussion:

$$10 \log (I_t/I_\infty) = 10 \log[1/(1-e^{-t/\tau})] \qquad \text{Eq. 1}$$

where I_t is the signal power (e.g., in watts) required for detection of a tone of duration t, I_∞ is the power required for detecting the same tone of indefinitely long duration, and τ is the time constant.

Equation 1 models sensory processes that occur up to a few hundred milliseconds after the onset of stimulation. The time constant (τ) of Equation 1 represents the amount of time that must pass before near-steady-state conditions are reached. Within this time period, a doubling of signal duration improves detection performance by approximately 3 decibels. In other words, 3 dB less signal to noise ratio is needed in this temporal regime to detect a signal twice as long as another.

For prolonged stimulation (i.e., signal durations much longer than τ), cognitive rather than sensory factors govern the

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[2] Zwislocki, J. J., THEORY OF TEMPORAL AUDITORY SUMMATION, Journal of the Acoustical Society of America, 32, 1046, 1960.

ability of human observers to make detection decisions. These nonsensory factors are discussed more fully on page 34 et seq. of this report. For reasons discussed more fully later, a doubling of signal duration in this temporal regime improves detection performance by only about 1.5 decibels.

Although most researchers subscribe to this general view, there is "remarkably little agreement about the duration of the ear's time constant" (Reference 3). Fidell et al. (Reference 3) provide the following brief summary of estimates of the ear's time constant: "One of the first to investigate the time constant of the ear was von Békésy (Reference 4). By varying the duration of an 0.800 kHz sinusoid, he concluded that about 180 msec was needed for it to reach its maximum loudness. Munson (Reference 5) used frequencies of 0.125, 1.000, and 5.650 kHz and estimated the time constant to be about 250 milliseconds. Miller (Reference 6) used broadband noise and estimated the time constant to be about 65 milliseconds at the higher sensation levels.

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[3] Fidell, Sanford, Karl S. Pearsons, Mario Grignetti and David M. Green, THE NOISINESS OF IMPULSIVE SOUNDS, Journal of the Acoustical Society of America, 48, No. 6, Part 1, 1970.

[4] von Békésy, G., ZUR THEORIE DES HORENS. UEBER DIE BESTIMMUNG DES EINEM REINEN TONEMPFINDEN ENTSPRECHENDEN ERREGUNGSEBIETES DER BASILARMEMBRAN VERMITTELS ERMUDUNGSERCHEINUNGEN, Phys. Z., 30, 115-125, 1929.

[5] Munson, W. A., THE GROWTH OF AUDITORY SENSATION, Journal of the Acoustical Society of America, 19, 584-591, 1947.

[6] Miller, G. A., PERCEPTION OF SHORT BURSTS OF NOISE, Journal of the Acoustical Society of America, 20, 160-170, 1948.

Small, Brandt, and Cox (Reference 7) performed a similar experiment, but found consistently smaller values. At high levels the time constant estimated by Small et al. appears to be in the range of 10-20 milliseconds. Von Port (Reference 8) investigated a variety of brief impulsive sounds. Most were gated noises of different bandwidths, center frequencies, and levels. He estimated that 70 milliseconds was a reasonable value of the time constant for a variety of stimuli. Von Niese (Reference 9) has summarized a number of studies from the Dresden Laboratories. He concluded that 25 milliseconds was a good average time constant value and argued that this estimate agreed nicely with reverberation time estimates. Von Zwicker (Reference 10) took issue with the conclusions of the Dresden group and presented more data from the Stuttgart Laboratory consistent with the longer estimate, approximately 100 milliseconds. Stevens and Hall (Reference 11), using a magnitude estimation technique, measured the time constant at about 150 milliseconds.

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[7] Small, A. M., Jr., J. F. Brandt, and P. G. Cox, LOUDNESS AS A FUNCTION OF SIGNAL DURATION, Journal of the Acoustical Society of America, 34, 513, 1962.

[8] von Port, E., UBER DIE LAUTSTARKE EINSELNER KURZER SCHALLIMPULSE, Acustica, 13, 211-223, 1963.

[9] von Niese, H., METHODE ZUR BESTIMMUNG DER LAUTSTARKE BELIEBIGER GERAUSCHE, Acustica, 15, 117-126, 1965.

[10] von Zwicker, E., EIN BEITRAG ZUR LAUTSTARKEMESSUNG IMPULSHALTIGER SCHALLE, Acustica, 17, 11-22, 1966.

[11] Stevens, J. C., and J. W. Hall, BRIGHTNESS AND LOUDNESS AS A FUNCTION OF STIMULUS DURATION, Perception and Psychophysics, 1, 319-327, 1966.

No one factor accounts for this variability in estimates of the ear's time constant, although factors such as experimental procedures, instructions to observers, and individual differences probably account for a fair amount of the variability. Watson and Gengel (Reference 12) contend that a major part of the variability may be attributed to genuine differences in the duration of the time constant as a function of frequency. Table 1, adapted from Reference 12, catalogs a number of studies from which Watson and Gengel conclude that "the form of the (temporal) integration function has been found to change with frequency, in that the time constant is reduced as frequency is increased in all experiments that satisfy the following conditions:

A) The range of signal durations investigated is wide enough to allow significant (as, 10-20 dB) differences between short-tone and long-tone thresholds;

B) A wide enough range of frequencies is investigated, generally, a range including at least the frequencies 250-4000 Hz; and

C) A relatively precise psychophysical procedure is used."

In summary, there is good reason to believe that the time constant of the ear is on the order of 200-300 milliseconds, and that it varies inversely with signal frequency. No information is available about the duration of the time constant at frequencies below 100 Hz. Watson and Gengel (Reference 12, Figure 6) present evidence that the time constant varies from about 150 milliseconds at 125 Hz to about 50 milliseconds at 8 kHz, a range of roughly 5 dB.

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[12] Watson, Charles S., and Roy W. Gengel, SIGNAL DURATION AND SIGNAL FREQUENCY IN RELATION TO AUDITORY SENSITIVITY, _The Journal of the Acoustical Society of America_, 46, No. 4, Part 2, 1969.

TABLE 1. SUMMARY OF STUDIES OF MONAURAL TIME-INTENSITY RELATION AT THRESHOLD, AS A FUNCTION OF FREQUENCY

	Signal Durations, msec	Background	Rise-Decay	Signal Frequencies, Hertz	Frequency Effect[a]
Hughes, 1946 (Ref. 13)	60 → 500	Quiet	Fast	250, 500, 1K, 2K, 4K	Definite
Garner, 1947 (Ref. 14)	1 → 100	Quiet	Fast	250, 1K, 4K, noise	No
Garner and Miller, 1947 (Ref. 15)	125 → 2000	Noise and Quiet	10 msec	400, 670, 1K, 1.9K	Possibly when replotted
Miskolczy-Fodor, 1953 (Ref. 16)	3 → 3000	Quiet	Not specified	250, 1K, 4K	Not for normals Yes for HIL's[b]
Eisenberg, 1956 (Ref. 17)	1 → 250	Quiet	Fast	500, 1K, 2K, 4K, 8K	Not systematic
Harris, Haines and Myers, 1958 (Ref. 18)	(Dependent variable)	Quiet	5 msec	250, 500, 1K, 2K, 3K, 4K, 6K, 8K	Not systematic (Short for HIL)
Plomp and Bouman, 1959 (Ref. 19)	1 → 5000	Noise	(Octave filter)	250, 500, 1K, 2K, 4K, 8K	Definite
Caspers, Lerche and Plath, 1960 (Ref. 20)	40 → 6000	Quiet	(? Filter)	100, 500, 1K, 5K	Definite
Simon, 1963 (Ref. 21)	12.5 → 400	Noise	(Elect. sw.)	1K, 4K	Yes, but more for HIL's
Zwicker and Wright, 1963 (Ref. 22)	10 → 500	Noise and Quiet	5 msec	500, 800, 1K, 1.2K, 1750, 3500	No
Elliott, 1963 (Ref. 23)	3 → 1000	Quiet	1 msec	500, 1K, 4K	Definite

[a]Studies in which definite frequency effect is found are those in which the integration time clearly decreases as a function of frequency.

[b]N: Listeners with normal audiogram; HIL: Hearing-impaired listeners.

TABLE 1. Continued.

	Signal Durations, msec	Background	Rise-Decay	Signal Frequencies, Hertz	Frequency Effect
Sheeley and Bilger, 1964 (Ref. 24)	0.25 → 256	Noise	Fast (0-crossing gate)	250, 1K, 4K	Definite
Olsen and Carhart, 1966 (Ref. 25)	10 → 1000	Quiet	7.5 msec	250, 1K, 4K, and noise	Clear effect below 50 msec
Wright, 1968 (Ref. 26)	10 → 500	Quiet	10 msec	125, 250, 500, 1K, 2K, 4K, 8K, and some others	Not for N Yes for HIL

[13] Hughes, J. W., THE THRESHOLD OF AUDITION FOR SHORT PERIODS OF STIMULATION, Proc. Roy. Soc. *(London)* B133, 486-490, 1946.

[14] Garner, W. R., THE EFFECT OF FREQUENCY SPECTRUM ON TEMPORAL INTEGRATION IN THE EAR, Journal of the Acoustical Society of America, 19, 805-815, 1947.

[15] Garner, W. R., and G. A. Miller, THE MASKED THRESHOLD OF PURE TONES AS A FUNCTION OF DURATION, Journal of Experimental Psychology, 37, 293-303, 1947.

[16] Miskolczy-Fodor, F., MONAURAL LOUDNESS BALANCE TEST AND DETERMINATION OF RECRUITMENT DEGREE WITH SHORT SOUND IMPULSES, Acta Otolaryngol, 43, 573-595, 1953.

[17] Eisenberg, R. B., A STUDY OF THE AUDITORY THRESHOLD IN NORMAL AND HEARING IMPAIRED PERSONS, WITH SPECIAL REFERENCE TO THE FACTORS OF THE DURATION OF THE STIMULUS AND ITS SOUND PRESSURE LEVEL, PhD Thesis, Johns Hopkins University, 1956.

[18] Harris, J. D., H. L. Haines and C. K. Myers, BRIEF TONE AUDIOMETRY, Acta Otolaryngol, 67, 699-713, 1958.

[19] Plomp, R., and M. A. Bouman, RELATION BETWEEN HEARING THRESHOLD AND DURATION FOR TONE PULSES, Journal of the Acoustical Society of America, 31, 749-758, 1959.

TABLE 1. Continued.

[20]Caspers, V. H., E. Lerche, and P. Plath, ÜBER DIE ABHÄNGIGKEIT DES HÄRSCHWELLENSCHALLDRUCKS VON DER TONDAUER BEI VERWEND UND VON KURZTÖNEN VERSCHIEDENER FREQUENZ, _Z. Laryngol. Rhinol. Otol._, 8-15, 1960.

[21]Simon, G. R., THE CRITICAL BANDWIDTH LEVEL IN RECRUITING EARS AND ITS RELATION TO TEMPORAL SUMMATION, _Journal of Auditory Research_, 3, 109-119, 1963.

[22]Zwicker, E., and H. N. Wright, TEMPORAL SUMMATION FOR TONES IN NARROWBAND NOISE, _Journal of the Acoustical Society of America_, 35, 691-699, 1963.

[23]Elliott, L. L., TONAL THRESHOLDS FOR SHORT DURATION STIMULI AS RELATED TO SUBJECT HEARING LEVEL, _Journal of the Acoustical Society of America_, 35, 578-583, 1963.

[24]Sheeley, E. C., and R. C. Bilger, TEMPORAL INTEGRATION AS A FUNCTION OF FREQUENCY, _Journal of the Acoustical Society of America_, 36, 1850-1857, 1964.

[25]Olsen, W. O., and R. Carhart, INTEGRATION OF ACOUSTIC POWER AT THRESHOLD BY NORMAL HEARERS, _Journal of the Acoustical Society of America_, 40, 591-599, 1966.

[26]Wright, H. N., CLINICAL MEASUREMENT OF TEMPORAL AUDITORY SUMMATION, _Journal of Speech and Hearing Research_, 11, 109-127, 1968.

RATIONALE FOR CURRENT INVESTIGATION

The purpose of the current study was to test a key assumption of a prior study undertaken for the Applied Technology Laboratory. The prior study (Reference 27) attributed differences in signal to noise ratio necessary to maintain constant detection performance at various frequencies to frequency domain effects. "Effective masking bandwidths" were estimated from the data of Fidell et al. (Reference 27) by assuming that all other things being equal, observed differences in signal to noise ratio necessary to maintain 76% correct detection performance in a two-alternative forced choice task in which observers [illegible] a sinusoid of various frequencies in noise could be attributed to differences in masking bandwidths at the various frequencies. This assumption is essential to Fletcher's (Reference 28) "critical ratio" argument used by Fidell et al. (Reference 27).

Because the 750 msec duration of the sinusoids detected in Reference 27 was much longer than the time constant of the ear, it was not possible to determine whether the frequency dependence of the ear's time constant affected the estimates of masking bandwidths inferred by the critical ratio method. The overall intent of the current research was to determine whether differences in integration time at different frequencies require revision of earlier estimates of masking bandwidths.

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[27] Fidell, S. A., R. D. Horonjeff and D. M. Green, LOW FREQUENCY ACOUSTIC DETECTION RESEARCH IN SUPPORT OF HUMAN DETECTION RANGE PREDICTION, USARTL-TR-79-25, U.S. Army Research and Technology Laboratories, Fort Eustis, Virginia, October 1979, AD A[illegible].

[28] Fletcher, H., SPEECH AND HEARING IN COMMUNICATION, New York, Van Nostrand, 1953.

The principal goals of the current study were therefore to extend existing information about the duration of the ear's time constant to lower frequencies, and to determine the relationship between integration time at these lower frequencies and at higher frequencies. This relationship must be known in order to assess the validity of the "all other things being equal" assumption that permitted inference of effective masking bandwidths in the frequency domain in Reference 27.

METHOD

PILOT STUDY

A pilot study was undertaken as a preliminary investigation of the effects of signal duration on detectability at various frequencies. The pilot study took advantage of an ongoing study of the symmetry of the auditory filter supported by the U.S. Army Tank-Automotive Research and Development Command (Reference 29). The goal of this study was to investigate the shape of the hypothetical auditory filter through which people are assumed to detect signals embedded in noise.

The logic of the TARADCOM study differed from that of Reference 27 in that masking bandwidths were measured directly rather than inferred by the critical ratio argument. The masking bandwidths were measured by forcing observers to listen for a sinusoid within a gap in the background noise spectrum that was discontinuous in the frequency domain. Digitally manufactured noise with precisely specified band limits and exceptionally steep skirts constrained the observers' listening strategies.

The trial procedure was a two-alternative forced choice detection task in blocks of 50 trials with a constant signal to noise ratio. Detection performance at each frequency was assessed at several different signal to noise ratios. Best fitting regression lines to these data were obtained for each observer and each frequency by calculating least square regression solutions to points in the linear portions of the psychometric functions.

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[29] Fidell, S. A., R. D. Horonjeff, S. R. Teffeteller, and D. M. Green, REVISION OF ACOUSTIC DETECTION RANGE PREDICTION MODEL BASED ON PSYCHOACOUSTIC STUDY OF LOW FREQUENCY MASKING, Bolt Beranek and Newman Report 4421, June 1980.

The slopes and intercepts of these regression equations were then averaged over all observers at each frequency to yield best fitting regression lines in order to estimate differences in signal to noise ratios necessary to maintain constant detection performance. All testing for the TARADCOM study was conducted at a single, fixed signal duration of 250 milliseconds.

The pilot study was conducted in conjunction with the TARADCOM study to provide an independent check on the critical ratio estimates of masking bandwidths of Reference 27. Observers were tested at additional signal durations of 100, 125, and 1000 milliseconds. All testing was conducted at 124 Hz with a continuous background noise spectrum. That is, the noise did not contain a discontinuous spectral notch in the frequency domain. Estimates of signal to noise ratios necessary to maintain constant detection performance at different signal durations were used as an independent check on estimates derived from the main study.

MAIN STUDY

Five observers (of 22.2 years average age) were paid at the rate of $3.50 per hour to investigate possible differences in detection performance caused by differences in integration time at different frequencies. (Observers in the previous low frequency acoustic detection study conducted for the Applied Technology Laboratory [Reference 27] detected sinusoids at 1000, 500, 250, 125, 63, and 40 Hz in a two-alternative forced choice detection task at a fixed signal duration of 750 milliseconds.) An incomplete factorial design of combinations of frequencies and durations was used in this study.

The range of signal durations was chosen from 50 to 1000 milliseconds. A priori information suggested that the integration time of the human auditory system is approximately 200-300 milli-seconds. Thus, three durations were chosen for testing on either side of this range. These durations included 50, 100, 200, 500, 750. and 1000 milliseconds.

Signal durations shorter than 50 milliseconds were avoided for two reasons. First, as noted by Garner and Miller in 1947 (Reference 15), rapid gating of a sinusoid at shorter durations spreads signal energy into spectral regions higher than the fundamental frequency of the sinusoid, thus complicating analyses of temporal domain effects. Second, signals with durations shorter than 50 milliseconds are essentially impulses, whose detectability may not be well predicted by the steady-state equations built into the Applied Technology Laboratory's range prediction software (Reference 30).

Table 2 contains the combinations of sinusoids and durations heard by the five observers in the main study. The order in which observers encountered sinusoids of varying durations was randomized, except at 40 Hz. Data were collected last at this frequency due to delays for equipment repairs and recalibration of the sound reproduction system.

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[30] Abrahamson, A. Louis, CORRELATION OF ACTUAL AND ANALYTICAL HELICOPTER AURAL DETECTION CRITERIA, Vol. 1, USAAMRDL-TR-74-102A, U.S. Army Air Mobility Research and Development Laboratory, Fort Eustis, Virginia, January 1975, AD B002071.

TABLE 2. SINUSOIDAL SIGNAL FREQUENCIES, BACKGROUND NOISES AND DURATION CONDITIONS TESTED

Frequency Duration (ms)	Narrow Band Background Noise			Wide Band Background Noise		
	40 Hz	63 Hz	125 Hz	125 Hz	250 Hz	1000 Hz
50	X	X		X	X	X
100	X	X		X	X	X
200	X	X	X	X	X	X
500	X	X		X	X	X
750	X	X	X	X	X	X
1000	X	X		X	X	X

Sinusoids at 500 Hz were not included in the detection task of the current study although sinusoids at this frequency were included in the prior study for the Applied Technology Laboratory (Reference 27). Sinusoids at 500 Hz were omitted since the data collected at 125, 250 and 1000 Hz varied by only a small degree. Preliminary findings suggested that the greatest difference in integration time of the human auditory system would be observed at low frequencies. Thus, greater emphasis was placed on detection tasks below 125 Hz.

New software was developed to provide trialwise adaptive adjustment of signal presentation levels. Trial-by-trial adjustment of signal to noise ratios was adopted because of its greater efficiency relative to blockwise adjustment of signal to noise ratios. In the prior study conducted by BBN for the Applied Technology Laboratory (Reference 27), data were collected under 32 test conditions (combinations of signal frequencies and durations); thus a more efficient means of collecting data was required.

In the trialwise-adaptive two-alternative forced choice signal detection procedure, the level of the signal can be adjusted after each detection decision. The method was drawn from the family of block up-and-down methods (Reference 31) in which the level of the signal is adjusted after a subset of trials. For example, in the "2 down, 1 up" procedure, the computer lowers the signal to noise ratio following each incorrect decision. The program can also look for a "3 down, 1 up" subset of trials in which three correct detection judgments are required at a given signal to noise ratio before lowering the level of the signal. The "2 down, 1 up" procedure estimates 71 percent correction detection performance from the test subject while the "3 down, 1 up" method estimates 79 percent correct detection.

The current experiment was set up to define a 76 percent correct level. To obtain this level of performance, the computer initially waited for two correct responses to lower the level of the signal. Thereafter, the computer randomly chose between two and three correct responses before lowering the level of the signal. Further details on the experimental operating procedures may be found in Appendix A along with sample line printer and oscilloscopic outputs.

Three background noise spectra were used in the study. Attention was once again paid to the influence of absolute level on detectability since Stevens (Reference 32) suggested "the critical duration (or endpoint of energy integration) appears

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[31] Levitt, H., and L. R. Rabiner, USE OF A SEQUENTIAL STRATEGY IN INTELLIGIBILITY TESTING, Journal of the Acoustical Society of America, 42, No. 3, 1967.

[32] Stevens, S. D. G., AUDITORY TEMPORAL INTEGRATION AS A FUNCTION OF INTENSITY, Journal of Sound and Vibration, 30(1), 109-126, 1973.

to become shorter with increasing overall level".

Table 3 shows the spectrum levels of the Gaussian masking noise used in this study. Sinusoids at 1000, 250 and 125 Hz were detected in noise at a spectrum level of 40 dB. Sinusoids at 125 and 40 Hz were detected in noise at a spectrum level of 60 dB, and tones at 63 Hz were detected in noise at a spectrum level of 55 dB. Table 3 also shows the assumed masking bandwidths at each frequency, derived from the data of the previous study conducted for the Applied Technology Laboratory (Reference 27).

TABLE 3. MASKING NOISE SPECTRUM LEVELS

Tone Frequency (Hz)	Assumed Masking Bandwidth (Hz)	Spectrum Level (dB re: 20μPa)	Masking Noise Bandwidth (Hz)	Masking Noise Over-all SPL (dB re: 20μPa)
40	90	60	17-400	85.8
63	66	55	17-400	80.8
125	70	60	17-400	85.8
125	70	40	17-2000	73.0
250	66	40	17-2000	73.0
1000	153.3	40	17-2000	73.0

*Taken from Reference 27, Table III, Page 45.

Observers detected the sinusoids in blocks of two-alternative forced choice trials. An individual trial contained two observation intervals of a duration specified by the experimenter. The observation intervals were separated by 500 milliseconds. A response interval of 750 milliseconds followed the second observation interval.

During one of the two observation intervals, a sinusoid at the tone frequency was embedded in the continuous background noise. Observers were instructed to press one of two lighted switches at this time to indicate the observation interval in which the sinusoid was thought to have occurred. If a response was made during the response interval, feedback was supplied to the observer by lighting the response switch corresponding to the interval in which the tone actually occurred. An intertrial period of 500 milliseconds separated successive trials.

To encourage conscientious performance, a bonus was awarded for each block of trials based on the observer's detection performance relative to the average detection performance at a given frequency and duration. Observers were tested in daily [illegible]-hour sessions with rest breaks approximately every 20 minutes. Testing continued over a period of several weeks under free-field listening conditions in an anechoic chamber. Figure 1 is a block diagram of the equipment used to conduct the experiment. Instructions to test subjects may be found in Appendix A.

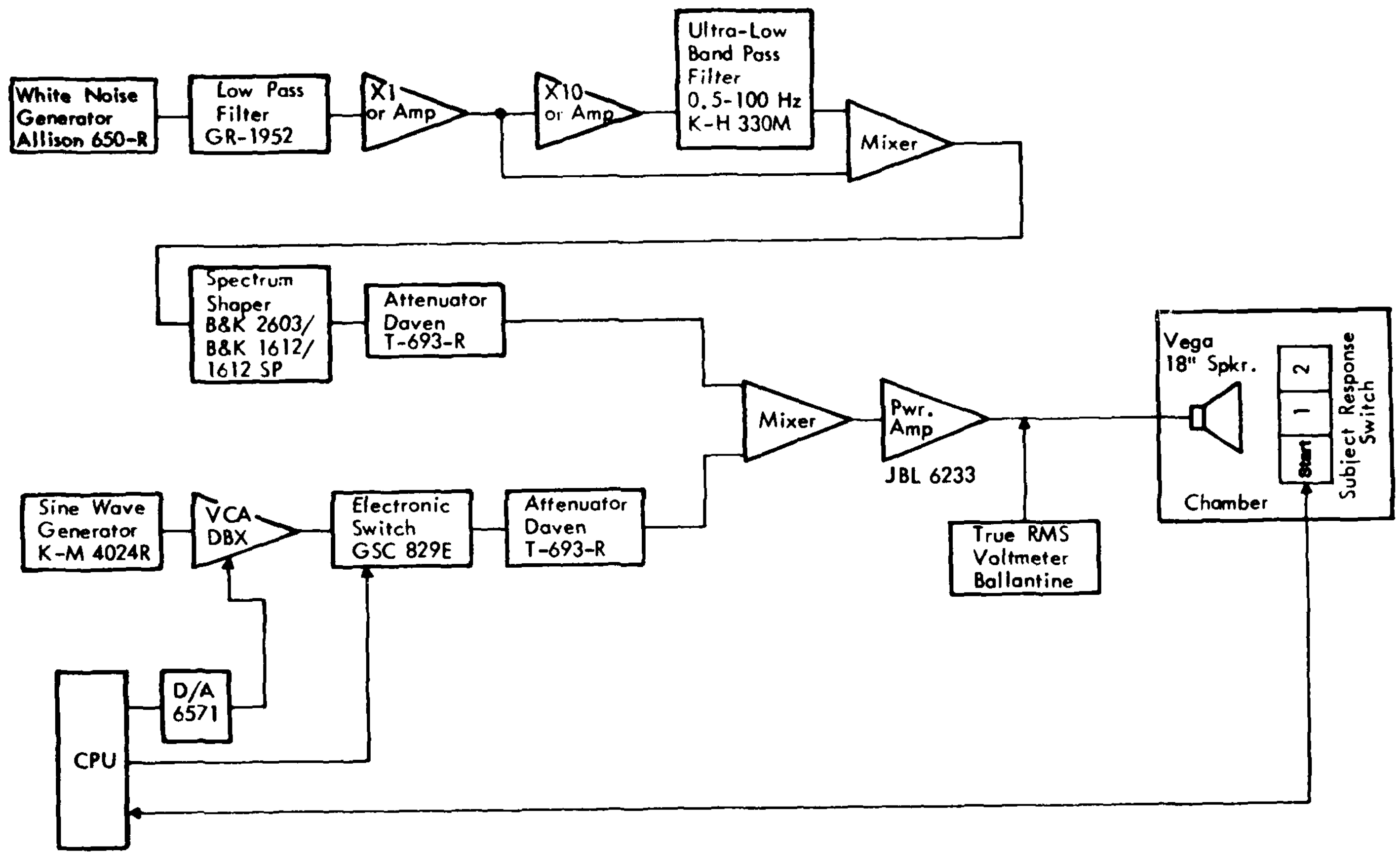

Figure 1. Block Diagram of Circuitry for Experiment.

RESULTS

PILOT STUDY

Preliminary data used to estimate the signal to noise ratios necessary for 76 percent correct detection were obtained at four different durations of sinusoids at 124 Hz, from 100 to 1000 milliseconds. Psychometric functions were estimated for each duration over a range of signal to noise ratios. The signal to noise ratios were adjusted to the performance of each observer. As a rule, signal to noise ratios slowly decreased over successive blocks of trials to avoid a sudden degradation in detection performance.

Signal to noise ratios were generally lowered in 1-dB steps until performance fell below 90 percent correct detection. Further adjustments were made in 0.5-dB steps. However, it was frequently necessary to repeat signal to noise ratio conditions several times to obtain stable estimates of performance at a given level, and least square regression lines at each frequency and signal duration were frequently based on a small number of data points due to rapid degradation in detection performance by observers. Thus, the major finding of the pilot study was the need for a more efficient psychophysical procedure to be used in the main study.

MAIN STUDY

The signal to noise ratios* for 76 percent correct detection performance (corresponding to d' = 1) averaged across the five observers are shown in Figures 2 through 6. Table 4 shows results for individual observers. The signal to noise "ratios"

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*See Appendix C.

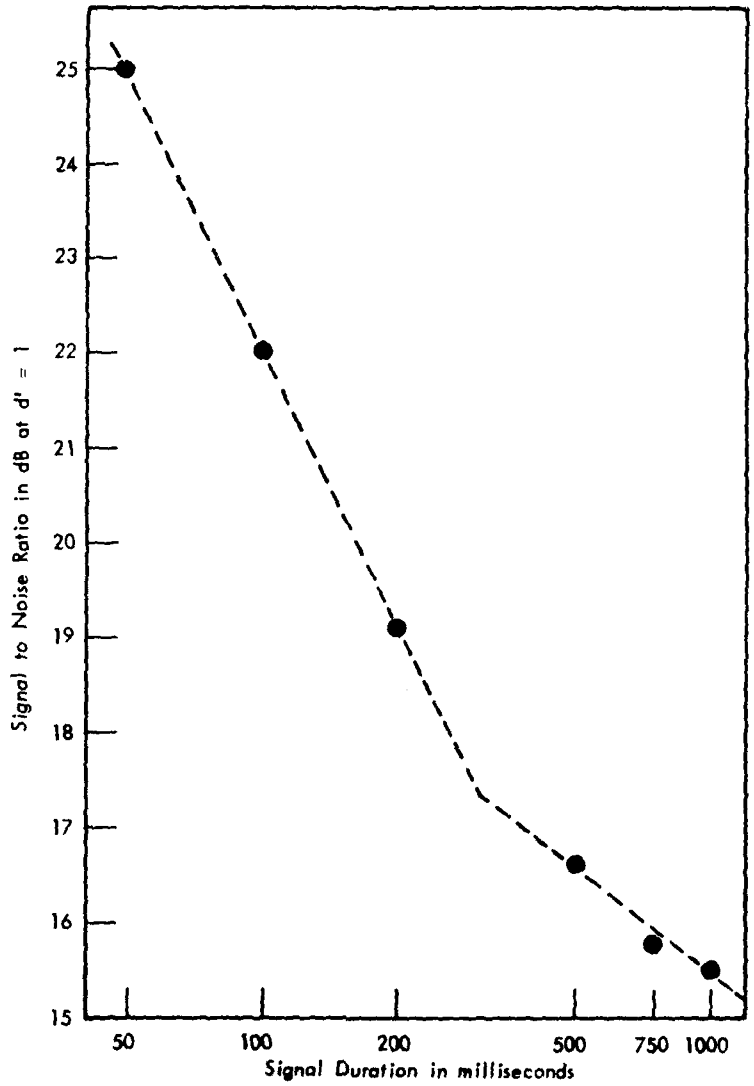

Figure 2. Average Signal to Noise Level at 76% Correct Detection With Varying Signal Durations at 40 Hz.

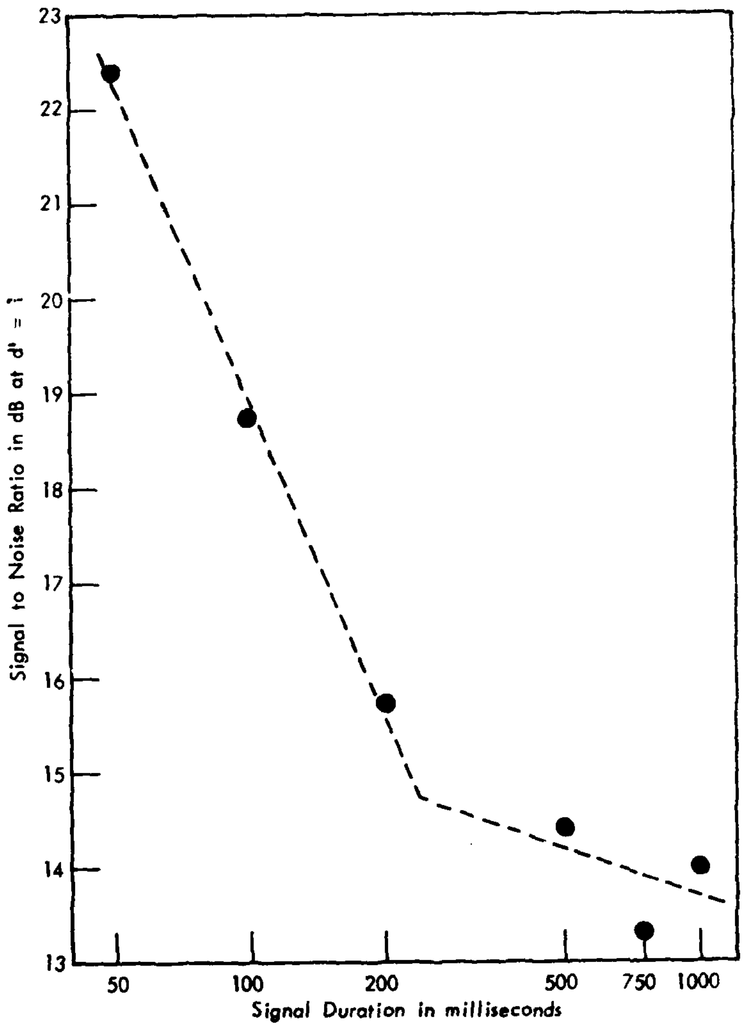

Figure 3. Average Signal to Noise Level at 76% Correct Detection With Varying Signal Durations at 63 Hz.

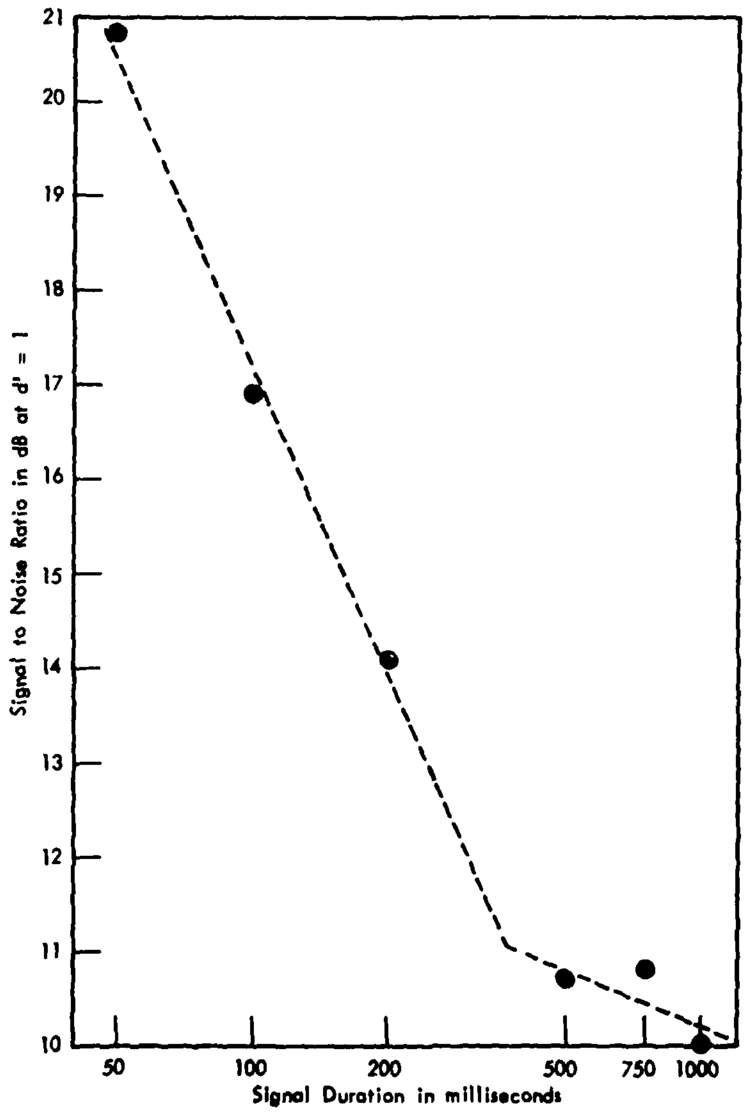

Figure 4. Average Signal to Noise Level at 76% Correct Detection With Varying Signal Durations at 125 Hz.

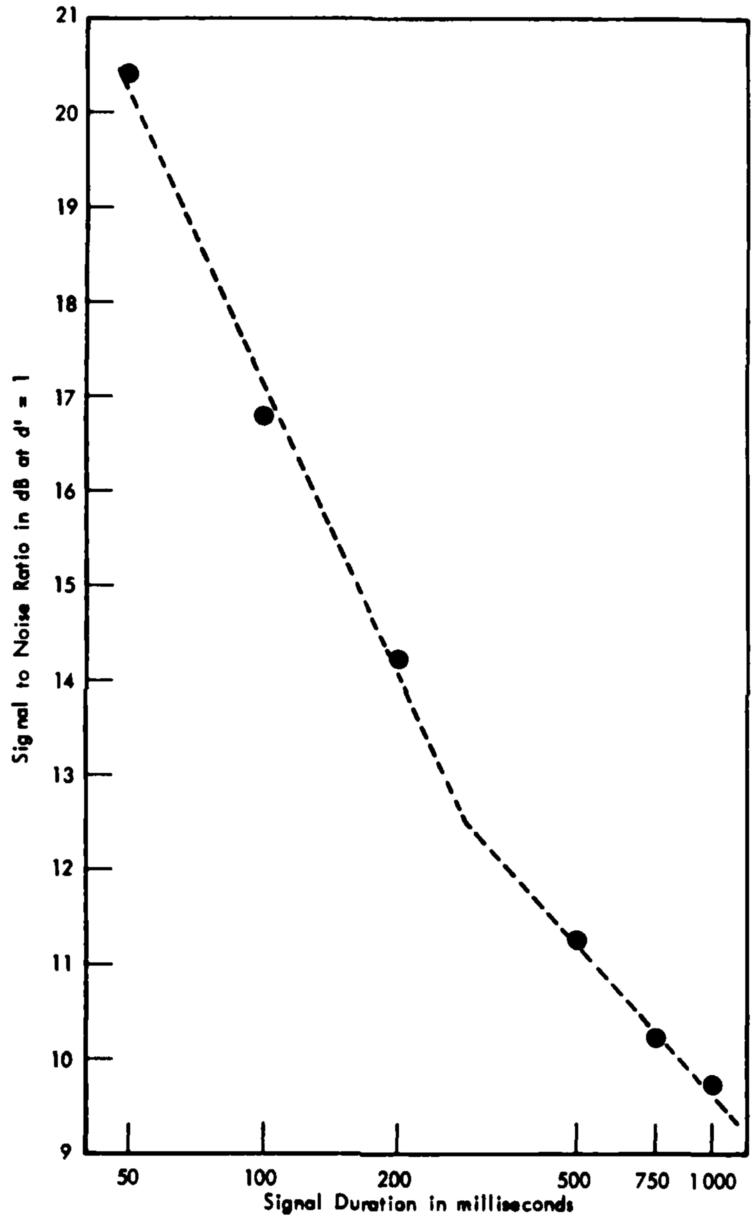

Figure 5. Average Signal to Noise Level at 76% Correct Detection With Varying Signal Durations at 250 Hz.

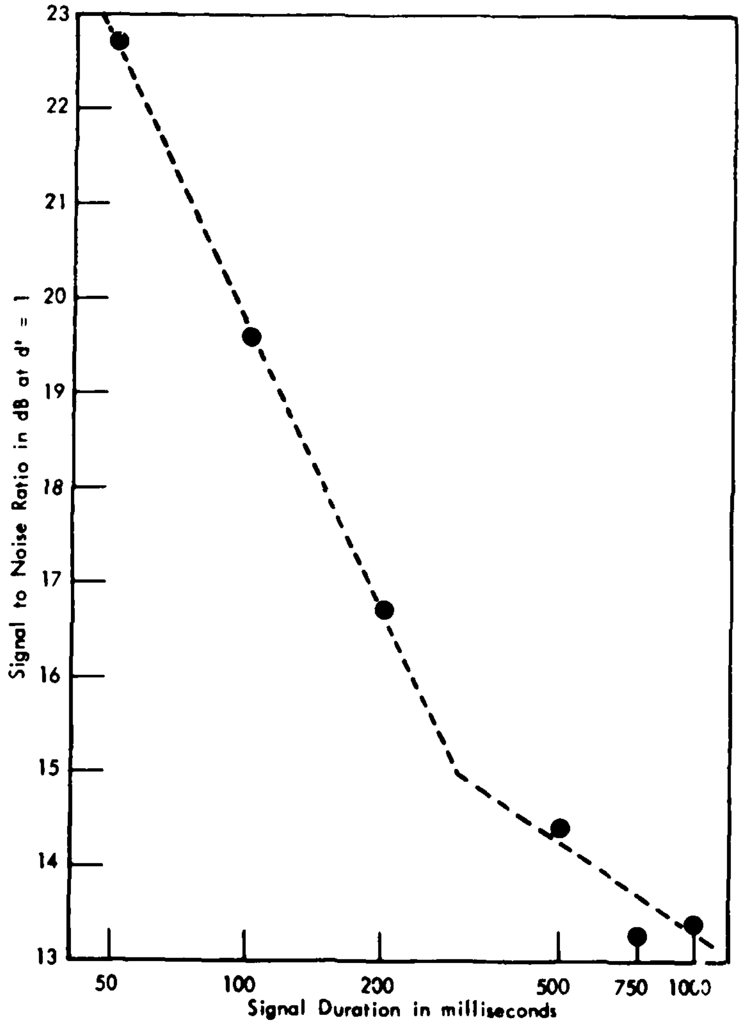

Figure 6. Average Signal to Noise Level at 76% Correct Detection With Varying Signal Durations at 1000 Hz.

TABLE 4. SIGNAL TO NOISE RATIOS FOR 76% CORRECT DETECTION PERFORMANCE FOR ALL OBSERVERS

40 Hz — Duration in msec

	50	100	200	500	750	1000
Subj. 1	--	22.6	19.2	16.1	15.1	14.3
2	24.2	21.6	18.7	16.8	15.9	15.8
3	24.4	21.3	18.7	15.6	14.7	14.3
4	27.6	24.6	21.2	20.3	19.6	19.6
5	23.7	19.7	17.9	14.4	13.9	13.7
$\bar{X}$	25.0	22.0	19.1	16.6	15.8	15.5
σ	1.77	1.81	1.24	2.23	2.22	2.40

63 Hz — Duration in msec

	50	100	200	500	750	1000
Subj. 1	[illegible]	19.4	15.9	14.7	14.1	14.4
2	22.2	19.3	[illegible]	14.5	[illegible]	13.9
3	--	--	--	--	--	--
4	22.5	[illegible]	15.9	[illegible]	13.8	14.6
5	[illegible]	17.7	[illegible]	14.1	12.6	[illegible]
$\bar{X}$	22.4	18.7	15.7	14.4	13.3	[illegible]
σ	.79	.79	.36	.25	[illegible]	.71

125 Hz (In Wideband Background)

	50	100	200	500	750	1000
Subj. 1	20.0	17.2	15.5	11.6	11.6	10.1
2	21.7	16.8	13.7	9.6	10.3	8.5
3	20.3	16.6	13.8	11.5	11.4	10.1
4	20.7	17.4	14.6	11.7	11.9	12.5
5	20.6	16.7	13.0	9.0	8.8	8.8
$\bar{X}$	20.7	16.9	14.1	10.7	10.8	10.0
σ	.64	.34	.96	1.28	1.27	1.58

125 Hz (In Narrowband Background)

	200	750
Subj. 1	15.5	12.8
2	15.3	9.0
3	15.9	13.0
4	16.5	14.9
5	13.9	11.3
$\bar{X}$	15.4	12.2
σ	.97	2.20

250 Hz

	50	100	200	500	750	1000
Subj. 1	[illegible]	[illegible]	14.9	11.3	9.9	[illegible]
2	20.4	16.5	14.5	[illegible]	[illegible]	[illegible]
3	[illegible]	17.1	[illegible]	[illegible]	[illegible]	[illegible]
4	[illegible]	[illegible]	13.4	[illegible]	[illegible]	[illegible]
5	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]
$\bar{X}$	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]
σ	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]

1000 Hz

	50	100	200	500	750	1000
Subj. 1	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]
2	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]
3	--	--	--	--	--	--
4	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]
5	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]
$\bar{X}$	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]
σ	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]

are based on the sound pressure level of the sine wave at each frequency minus the average spectrum level (dB per Hertz) of the noise over the estimated masking bandwidths of Reference 27 (see Table 3). Notice that as signal durations at all frequencies increase, the signal to noise ratios required for a constant level of detection performance decrease.

Figures 2 through 6 show two least square regression lines. The first regression line is drawn through the three shortest duration points that fall below the expected integration time of the ear. The second line is drawn through the three points presented at durations longer than the expected integration time. In all cases, the signal to noise ratio for 76 percent correct detection performance decreases as the duration of the signal increases.

Effective masking bandwidths were estimated for six frequencies with signal durations of 750 milliseconds in the previous study for the Applied Technology Laboratory (Reference 27). Sinusoids presented at 750 milliseconds were included in the current study along with five other durations of signal in order to compare the results of the two studies. Figure 7 compares the average signal to noise ratios when d' = 1 with a signal duration of 750 milliseconds for the previous and current studies for the Applied Technology Laboratory.

Figure 7 also shows the 95 percent confidence interval about the mean when d' = 1 at each frequency. As can be seen from Figure 7, the differences at higher frequencies (above 125 Hz) are unlikely to have arisen by chance alone. No simple explanation can be provided for the differences at the higher frequencies. However, greater consideration should be given to the mean values found in the current study, due to the increased precision afforded by the psychophysical procedure used in the current study.

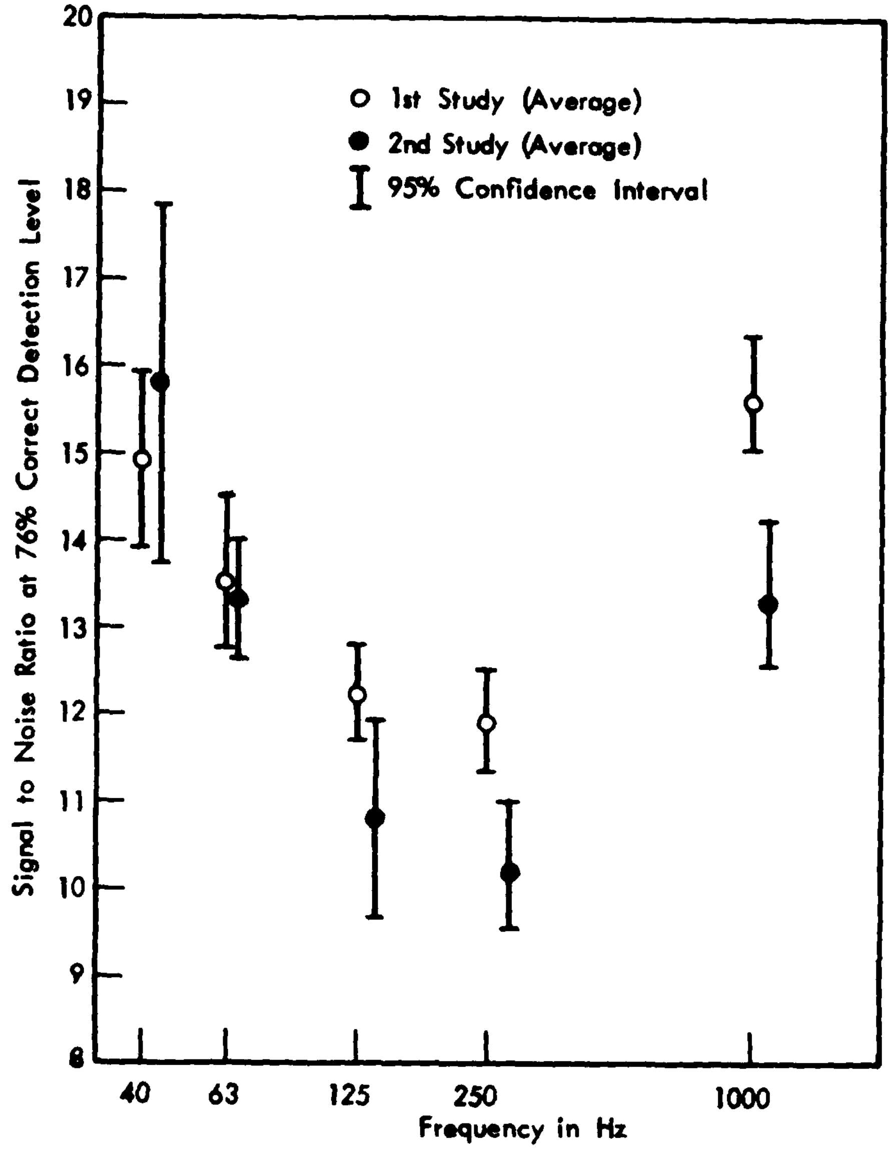

Figure 7. Comparison of Signal to Noise Ratios at 76% Level at Different Frequencies During 1st Study and 2nd Study With 750 Millisecond Duration.

The effect of the duration of the signal, which was of most interest in the study, was examined by plotting the signal to noise ratios as functions of signal frequency and duration. The expectation was that at shorter signal durations, signals of equal energy (defined as the signal sound pressure level plus 10 times the logarithm of the signal duration) are equally detectable.

Figure 8 shows regression lines through the signal to noise "ratios" corresponding to d' = 1 points averaged across the five observers. The metric of the ordinate, E/N_o, is the signal energy in decibels (sound pressure level plus 10 times the log of the duration of the signal in seconds) minus the spectrum level of the noise in decibels (sound pressure level in a 1-Hz wide band). Table 5 contains the E/N_o values at the 76 percent correct detection level for all observers at five frequencies. These values differ from those in Table 4 only by the 10 log duration correction. The E/N_o values for each observer plotted as functions of the log of the signal duration may be found in Appendix B.

The data collected at signal durations of 50, 100 and 200 milliseconds suggest that detection performance with signal durations shorter than the auditory integration time depends on the signal energy alone. A horizontal line is drawn through the average of the three data points in Figure 8 at the shortest durations. The mean deviation about the horizontal line is only 0.25 dB for all frequencies.

For signal durations in excess of the aural integration time, prior research, along with mathematical models of the detection process, suggested that detection performance is based on a vector summation of the signal detectability (d') in sequential nonoverlapping integration periods. That is, if the signal duration is longer than the aural integration time, the listener

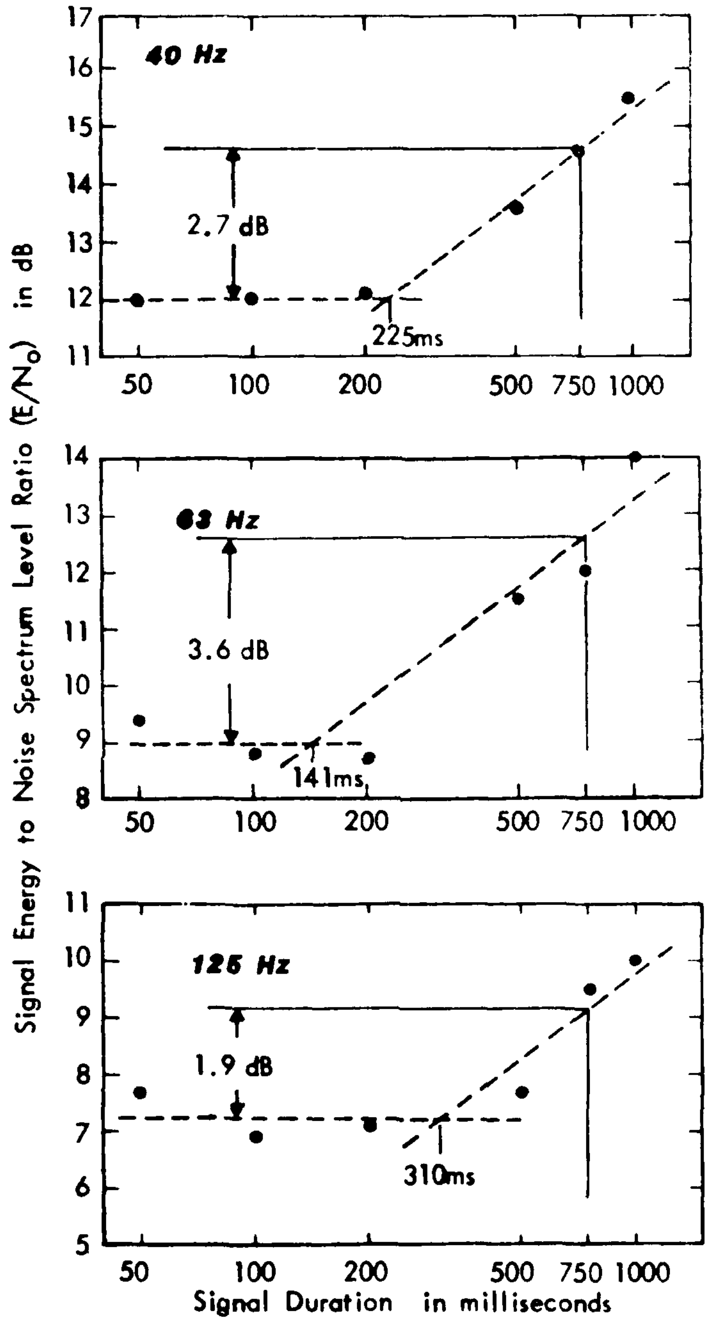

Figure 8. E/N_o Vs Duration Curves for Five Frequencies.

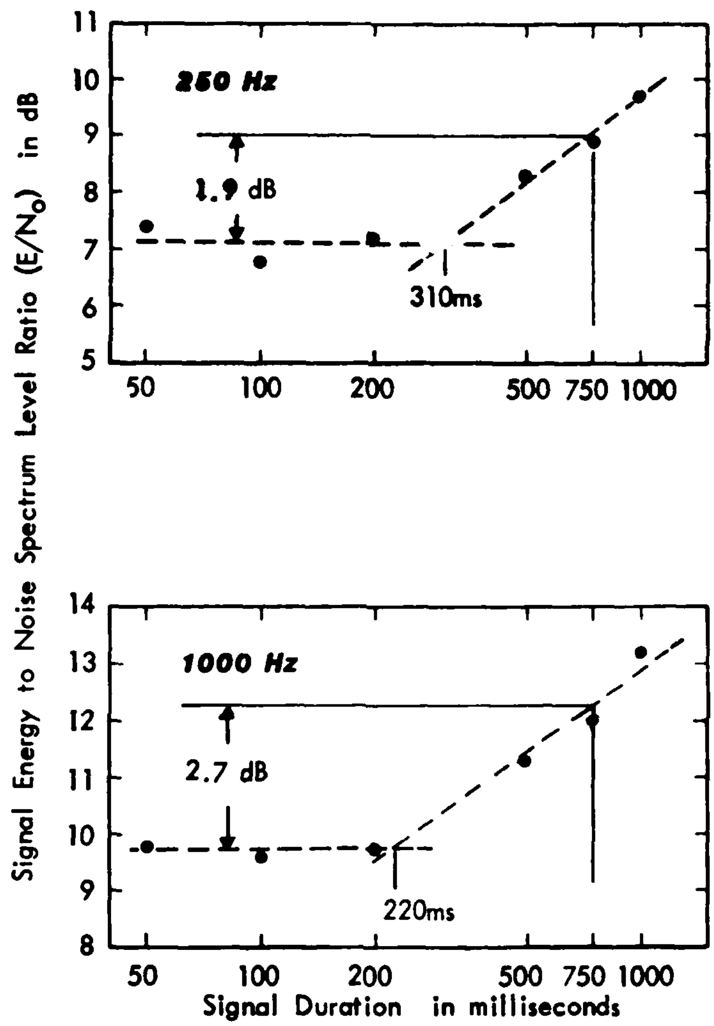

Figure 8. Continued.

TABLE 5. E/N_o VALUES AT 76% CORRECT LEVEL FOR ALL OBSERVERS

40 Hz — Duration in msec

	50	100	200	500	750	1000
Subj. 1	--	12.6	12.2	13.1	13.8	14.3
2	11.2	11.6	11.7	13.8	14.6	15.8
3	11.4	11.3	11.7	12.6	13.4	14.3
4	14.6	14.6	14.2	17.3	18.4	19.6
5	10.7	9.7	10.9	11.4	12.7	13.7
$\bar{X}$	12.0	12.0	12.1	13.6	14.6	15.5
σ	1.77	1.81	1.24	2.23	2.24	2.40

63 Hz — Duration in msec

	50	100	200	500	750	1000
Subj. 1	10.4	9.4	8.9	11.7	12.8	14.4
2	9.2	9.3	9.0	11.5	11.3	13.9
3	--	--	--	--	--	--
4	9.5	8.6	8.8	11.5	12.5	14.6
5	8.5	7.7	8.2	11.1	11.3	13.0
$\bar{X}$	9.4	8.8	8.7	11.5	12.0	14.0
σ	.79	.79	.36	.25	.79	.71

125 Hz (*In Wideband Background*)

	50	100	200	500	750	1000
Subj. 1	7.0	7.2	8.5	8.6	10.3	10.1
2	8.7	6.8	6.7	6.6	9.0	8.5
3	7.3	6.6	6.8	8.5	10.1	10.1
4	7.7	7.4	7.6	8.7	10.6	12.5
5	7.6	6.7	6.0	6.0	7.5	8.8
$\bar{X}$	7.7	6.9	7.1	7.7	9.5	10.0
σ	.64	.34	.96	1.28	1.27	1.58

125 Hz (In Narrowband Background)

	200	750
Subj. 1	8.5	11.5
2	8.3	7.7
3	8.9	11.7
4	9.5	13.6
5	6.9	10.0
$\bar{X}$	8.4	10.9
σ	.97	2.20

250 Hz

	50	100	200	500	750	1000
Subj. 1	7.8	7.0	7.9	8.3	7.7	10.4
2	7.4	6.5	7.5	7.5	9.5	9.7
3	7.2	7.1	7.0	8.4	9.4	9.8
4	7.7	7.3	6.4	9.2	9.6	9.9
5	6.7	6.2	7.1	7.9	8.5	8.8
$\bar{X}$	7.4	6.8	7.2	8.3	8.9	9.7
σ	.44	.45	.56	.63	.82	.58

1000 Hz

	50	100	200	500	750	1000
Subj. 1	10.5	10.6	9.2	11.5	12.7	13.7
2	9.6	8.7	9.8	10.8	12.0	13.0
3	--	--	--	--	--	--
4	9.1	9.0	10.4	12.7	12.6	13.6
5	9.8	10.2	9.3	10.8	10.8	13.4
$\bar{X}$	9.8	9.6	9.7	11.4	12.0	13.4
σ	.58	.92	.55	.90	.87	.31

will (1) divide the total signal duration into N segments, each of which is only one integration period long, (2) compute the signal detectability separately for each segment, and (3) compute the composite detectability for the entire signal by

$$d'_c = [(d'_1)^2 + (d'_2)^2 + \ldots (d'_N)^2]^{1/2}$$

Under such a detection model the signal energy required for equal detectability increases at the rate of 5 dB per multiple of 10 times duration. A best fitting regression line of this slope is shown for the data at the 500, 750 and 1000 millisecond durations at each frequency. The intersection point of the two line segments provides the best estimate of the aural integration time at each frequency. At 63 Hz the aural integration time appears to be less than that at 125 Hz and above. However, the estimate of integration time at 40 Hz provided by the intersection of the two line segments is almost equal to the integration time at 1000 Hz.

The difference in E/N_o values for detection of signals of 750 milliseconds duration (the duration used in Reference 27) and for those within the constant energy regime was of central interest in this study.

Table 6 shows the ΔE values, or the difference in E/N_o values found with a 750 millisecond signal duration and the average E/N_o value over the 50, 100, and 200 millisecond durations (the equal energy regime). The average ΔE value is also shown for each frequency in Figure 8. Note that at 125, 250 and 1000 Hz the mean differences are relatively small (2.2 dB). Deviations about this mean were only 0.5 dB. Using Fletcher's critical ratio, such small differences produce only minor changes in masking bandwidths estimated by the critical ratio assumption.

The largest difference occurred at 63 Hz, where a 3.6 dB difference from the E/N_o value found at 750 milliseconds duration and the constant energy average was observed. However, at 40 Hz the difference was 2.7 dB, identical to the difference observed at 1000 Hz.

TABLE 6. ΔE VALUES AS A FUNCTION OF FREQUENCY, IN dB

	Subject						
Frequency	1	2	3	4	5	$\overline{X}$	σ
40	1.4	3.4	1.7	4.5	2.3	2.7	1.2[illegible]
63	3.5	3.2	-	4.0	3.8	3.6	0.3[illegible]
125	2.2	0.8	2.8	3.2	0.8	2.0	1.1[illegible]
250	1.3	1.9	2.2	2.5	1.8	1.9	0.4[illegible]
1000	2.6	2.5	-	3.5	1.9	2.7	0.6[illegible]

Figure 9 shows the average ΔE values found at the five frequencies and the 95 percent confidence interval around each mean value. The lack of systematic differences in ΔE values over the frequency range of interest suggests the absence of meaningful duration effects as a function of frequency.

Sinusoids of 125 Hz were presented at all six durations in a background noise spectrum level of 40 dB. The same signals were also detected at 200 and 750 millisecond durations in a 60 dB background noise spectrum level. The average difference in detection performance observed between the two noise level conditions was 1.4 dB. A difference of 1.6 dB was found between these two background spectrum levels in Reference 28. The excellent agreement of these two findings substantiates the effect of absolute level of the background noise on detection performance.

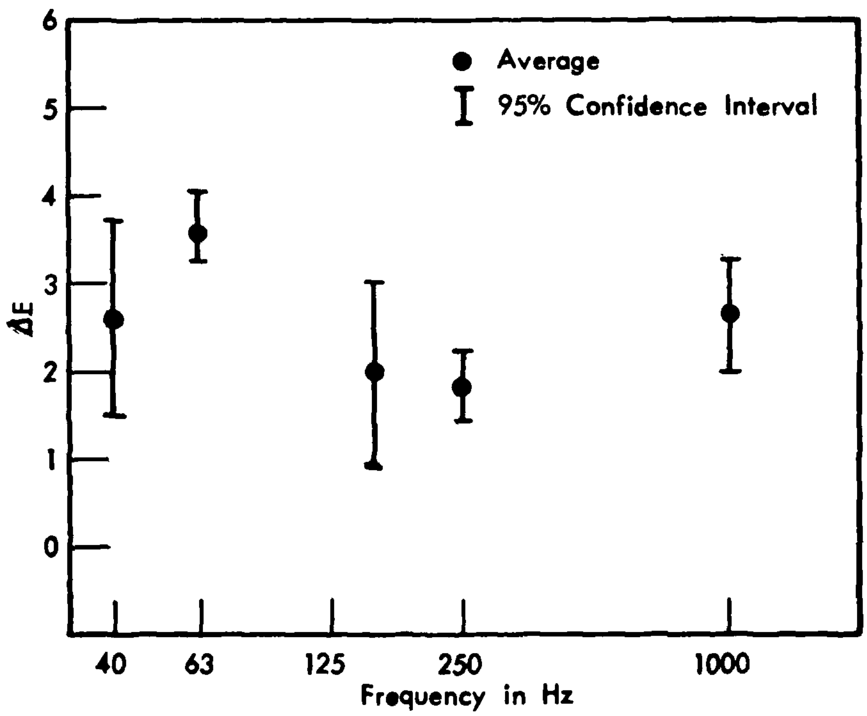

Figure 9. Average ΔE and the 95% Confidence Interval as a Function of Frequency.

DISCUSSION

The orderliness of the data shown in Figures 2 through 6 reflects the precision of the psychophysical procedure employed in this study. For signal durations less than the aural integration time, data points never deviate more than 0.25 decibels from the least square regression line. For the higher signal durations, the maximum deviation from the regression lines is less than 0.5 decibels. Considering that each data point was determined on the basis of approximately 1500 two-alternative trials, this regularity is not altogether surprising.

Initial estimates of the aural integration time from the literature (approximately 200-300 milliseconds) were, in retrospect, excellent for purposes of experimental planning. The three signal durations chosen on either side of this value fell decidedly into the temporal regimes (shorter and longer than the auditory integration time) discussed on pages 32 et seq. Durations at or near the integration time appear to have been successfully avoided as planned.

The dashed lines shown in Figure 8 are the best fits of the data to the detection model for the temporal regime greater than the auditory integration time. Note that the ordinate in Figure 8 is signal <u>energy</u>, not signal <u>level</u>. Under the model the signal energy required for equal detection performance should remain constant up to durations equal to the integration time. For greater durations, the required signal energy should increase at a rate of 1.5 dB per doubling of duration. The dashed lines of these slopes appear to fit the data quite well, with maximum deviations of 0.5 dB. The intersection of these two lines provides the best estimate of the aural integration time. Note that at all frequencies except 63 Hz, the estimates fall in the range of 220 to 310 milliseconds, but with no apparent systematic dependency on frequency. The exception, 63 Hz, actually estimates an integration time (141 milliseconds) which is in the constant energy regime and is in all probability unrealistically low. There is, unfortunately, no immediate explanation for this anomaly.

Upon more careful inspection of the 500, 750 and 1000 milliseconds data points in Figure 8, it appears that a slightly steeper slope (perhaps 2.0 to 2.3 dB per doubling) might provide a better fit to the data. This slope would alleviate the anomaly found at 63 Hz, but would require a reassessment of the assumed detection model at higher durations. To put this slope into some perspective, 1.5 dB/doubling provides the very best detection performance when the observer uses all of the information over the total signal duration with perfect recall of assessed detectability in each temporal subdivision. In contrast, 3.0 dB/doubling would be expected if the observer listened for only one integration period, made his judgment, and completely ignored the remainder of the signal duration. Thus, a slope of 2.0/doubling indicates that the observers are still able to perform the temporal subdivision process, but may either (1) forget or underrate the assessed detectability of the signal at its onset, or (2) become somewhat less vigilant in the detection task toward the end of the signal.

The comparison of signal to noise ratios in Figure 7 suggests non-negligible differences between results obtained in Reference 27 and the current study. With the exception of 40 and 63 Hz, data from the two studies do not fall within each other's 95% confidence intervals. Although the different trial procedures of the two studies might account for some of this difference, it seems unlikely that the differences would be frequency dependent. The increased precision of measurement of the psychophysical procedure used in the current study is thought to provide more reliable and stable results than those of Reference 27.

Table 7 compares estimated masking bandwidths for the current study and Reference 27. These estimates are based on application of Fletcher's critical ratio assumptions to the data of Table 3 (in the case of Reference 27) and Figure 7 (in the case of the current data). The two sets of estimates are in considerable disagreement at frequencies below 125 Hz.

TABLE 7. BANDWIDTHS OF HYPOTHETICAL AUDITORY FILTER INFERRED BY FLETCHER'S CRITICAL RATIO METHOD

Center Frequency (Hz)	Estimated Masking (Reference 27) (Hz)	Bandwidth Present Study (Hz)
1000	153*	153*
250	66	72
125	70	82
63	66	150
40	90	260

*Forced to agree with value assumed in Reference 30.

It seems clear that the low frequency masking bandwidth estimates of the present study are simply not credible. For the hypothetical auditory filter centered at 40 Hz to be 260 Hz wide, it would have to be far more asymmetric than it is known to be on the basis of other research (Reference 29). For that matter, the estimate of the masking bandwidth at 40 Hz from Reference 27 (90 Hz) is also implausibly wide.

The obvious problem is the assumed reference bandwidth of 153 Hz at 1000 Hz. This value was chosen to agree with the value incorporated in the acoustic detection range software described in

Reference 30. Abrahamson in turn adopted this value from the research of Greenwood (Reference 33). Patterson (Reference 34) has since shown that Greenwood's estimate is too large by a factor of at least two, due to inappropriate assumptions about the detection strategies used by the observers in Greenwood's experimentation.

A more generally accepted value for the effective masking bandwidth at 1000 Hz is about 65 Hz. Applying Fletcher's critical ratio assumptions to the current data with this reference value produces a masking bandwidth estimate of 35 Hz at 125 Hz, a figure in good agreement with the estimate of 28 Hz from Reference 29. Since no testing was conducted at frequencies below 125 Hz in Reference 29, it is not possible to determine how meaningful the wider bandwidth estimates obtained in the present study at 63 and 40 Hz may be.

These anomalies in the frequency domain, while meriting further study in their own regard, do not affect the overall conclusion that can be drawn from the current study in the temporal domain. There is no reason to modify the prediction model of Reference 30 on the basis of the empirical evidence of systematic differences in temporal integration intervals at low frequencies.

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[33]Greenwood, D. D., AUDITORY MASKING AND THE CRITICAL BAND, Journal of the Acoustical Society of America, 33, 484-502, 1961.

[34]Patterson, R. D., AUDITORY FILTER SHAPES DERIVED WITH NOISE STIMULI, Journal of the Acoustical Society of America, 59, 640-654, 1976.

CONCLUSIONS

1) The signal to noise ratio that human observers require to maintain a constant level of detection performance (percent correct decisions) decreases as the duration of the signal to be detected increases from 50 to 1000 milliseconds.

2) Detection performance of human observers listening for acoustic signals embedded in noise can be modeled as a simple energy integration for signal durations less than about a quarter of a second (τ in Eq. 1, p. 8 = 240 milliseconds).

3) For present purposes, the integration time of the human auditory system (τ) may be regarded as invariant over the frequency range of 40 to 1000 Hz. Thus, no modification to ATL's acoustic detection range prediction software is needed to account for differences in the short term (less than a quarter of a second) detectability of acoustic signals of different frequencies between 40 and 1000 Hz.

4) Low frequency masking bandwidths inferred from the data of the present study are wider than those inferred from the data of Ref. 27. Both sets of estimates are very sensitive to small changes ($\pm$ 1 dB) in signal to noise ratios needed to maintain constant detection performance at low frequencies.

RECOMMENDATIONS

Because ATL's acoustic detection range prediction software is insensitive to signal duration, it predicts the same detection range for both instantaneously and continuously audible signals. The results of the present experimentation clearly show that signals of greater duration require lower signal to noise ratios to be detected with the same probability as signals of shorter durations. Analyses of the present findings show two relationships between signal to noise ratios needed for constant detection performance as a function of signal duration:

1. For signal durations less than the integration time (τ), human detection may be modeled as a simple energy integration in which process signals of equal energy (product of amplitude and duration) are equally detectable. Signal levels required for constant detection performance therefore decrease by 3 dB for every doubling of duration.

2. For signal durations greater than the auditory integration time, human detection may be modeled as a temporal subdivision process in which detectability (d') is independently assessed in sequential, nonoverlapping integration periods and is orthogonally vector summed. Signal levels required for constant detection performance therefore decrease at a rate of 1.5 dB for every doubling of duration.

If incorporated in ATL's software, these relationships would improve the generality of its prediction algorithms.

APPENDIX A
TEST PROCEDURE

The trialwise adaptive two-alternative forced choice signal detection procedure begins with a conversation between the experimenter and the computer-based system. The system prompts the experimenter for information needed to conduct the experiment by inquiring about the conditions for data collection. The program begins by typing a one-line identifier and waiting for the experimenter to type one of three commands. A "C" directs the program to enter calibration mode. The experimenter is asked for the data and experimenter identification. The program then asks for specific information about the signal and the background noise and allows the experimenter to adjust the level of each. The experimenter adjusts the level of the signal to 20 dB above the level at which it is barely audible. When the calibration procedure is complete the program returns to command mode.

The second command, an "M", allows the experimenter to change the various operating parameters of the program. The program presents the current values of the parameters for confirmation and/or modification. The experimenter may change the three trial parameters which define the length (in msec) of the observation, intratrial and response intervals. The experimenter then specifies the maximum number of trials to be presented in a given block of trials, and the initial step size and the smallest step size in decibels. The experimenter can also request paper tape output.

Next, the experimenter specifies stopping criteria: the desired size of the 95% confidence interval in decibels and the maximum number of reversal cycles. The experiment terminates either when it reaches a criterion confidence interval or by completing a specified number of reversal cycles in responses.

The bonus parameters may also be altered by the experimenter. The experimenter defines the bonus in cents to be paid for an average performance by the test subject as well as the additional payoff for performance that falls above or below average. A differential payoff rate is calculated in cents per decibel deviation from average performance. The experimenter also limits the maximum bonus paid to the test subject. The program does not permit a negative bonus to be computed.

Following definition of the operating parameters, the experimenter proceeds to actual running of the program by typing an "R" for Run. The experimenter enters the subject identification and specifies a starting signal-to-noise ratio in dB. The program can continue to operate automatically for a number of runs as determined by the experimenter. The experimenter's last input to the program is signal to noise ratio at the detection level of interest. A heading is printed on the teletype for information supplied to the experimenter following each block of trials.

Finally, the experimenter types a 'space' to start the experiment. This causes the "Start" button on the subject's response box to light up. After depression of the "Start" button, a grid on a storage oscilloscope is plotted and the experiment commences. This display provides a real-time plot of two experimental parameters: the level of the signal and the performance of the subject. Figure A-1 shows a sample of the grid displaying the level of the signal on each of the two-alternative forced choice trials. For each signal presentation, the display indicates

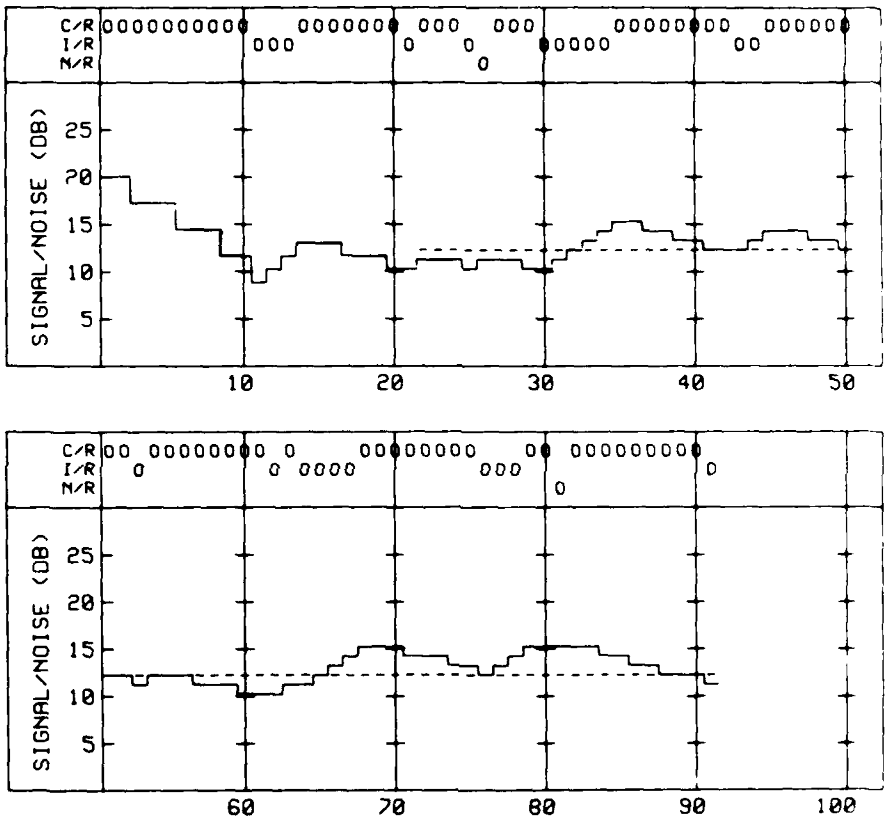

Figure A-1. Sample of Output on Oscilloscope Display.

whether the response was correct - "C/R", incorrect - "I/R", or that there was no response - "N/R" by placing a circle in the proper area of the grid. The signal level for the next trial is displayed on the grid just before it is presented to the test subject. The performance of the subject is shown at the top of the grid.

The oscilloscopic display shows the level of the signal and the response of the subject for the entire block of trials. The signal is initially presented at a very high level so that it is clearly audible. Adjustments to the signal to noise ratio are initially large, but are halved after the first reversal of direction of adjustment until a minimum step size criterion is reached.

After two complete reversal cycles the program checks to determine if it has reached a minimum step size as specified by the experimenter. If the current step size equals the minimum step size, the computer considers the first two reversal cycles as training and proceeds to cumulate statistics of detection performance. If the step size is above the minimum step size, the computer continues with another cycle of training trials until a reversal is reached. A minimum of two training cycles is required at the beginning of each block.

The first 21 trials, shown in Figure A-1, were treated as training trials. Following two reversal cycles the computer began cumulating response data. The subject completed 7 reversal cycles in the remaining 70 trials of the block. The computer terminated the block after 91 trials when it determined that there were too few remaining trials in the block of 100 to complete another reversal cycle.

The experimenter may halt the experiment while the program is running in four ways: typing a "T" causes the experiment to stop after the next trial; typing a "C" causes the experiment to halt after completion of the next reversal cycle; typing a "B" causes the program to halt after the next block of trials; and typing an "R" halts the experiment after completion of the next run. The experimenter may cancel any of the operator intervention commands by typing a 'space' before the experiment halts. After the experiment stops, statistics cumulated up to the point of stopping for the block are printed on the teletype. However, the statistics are not cumulated for output on the line printer until the experimenter indicates whether to keep or discard the current block. The experimenter then elects either to continue or end the run.

Statistics are printed on the line printer for each reversal cycle. Figure A-2 shows an example of the output to the line printer for a block of trials. The output first indicates the number of training trials for each block. For each reversal cycle, the beginning and ending time are printed along with the mean signal to noise ratio for the cycle, the number of trials before a reverse response occurred and the number of trials in which the subject failed to make a response.

Cumulative statistics for the blocks are provided starting with the first reversal cycle. The mean signal to noise ratic, 95 percent confidence interval, and the percent correct in interval one and interval two are cumulated until the end of the block.

TRIALWISE ADAPTIVE TWO-ALTERNATIVE FORCED CHOICE
SIGNAL DETECTION PROCEDURE

SUBJECT: SAF
SIGNAL: 400 HZ AT 10HZ REP RATE
NOISE: GAUSSIAN

DATE: 05 MAY 1980
EXPERIMENTER: RDH
BBN JOB #: 08084

OBSERVATION INT'VL (MSEC) 750
INTRA-TRIAL INT'VL (MSEC) 500
RESPONSE INT'VL (MSEC) 1000

MAX STEP SIZE (DB) 2 5
MIN STEP SIZE (DB) 1 0

		CURRENT REVERSAL CYCLE				CUMULATIVE INFORMATION			
BEGIN	END	REV CYCLE	MEAN S/N	# OF TRIALS	MISS RESP	MEAN S/N	95% C I	%(C) INT1	%(C) INT2
11 25 27	11 27 54	(TRAINING)		25	0				
11 27 54	11 28 21	1	-11 0	9	0	-11 0	0 00	80 0	75 0
11 28 21	11 28 51	2	-11 5	7	0	-11 [illegible]	0 48	70 0	83 3
11 28 51	11 29 01	3	-12 0	3	0	-11 5	0 56	66 6	85 7
11 29 01	11 29 11	4	-12 0	3	0	-11 7	0 46	69 2	77 7
11 29 11	11 29 32	5	-11 0	6	0	-11 5	0 43	56 6	80 0
11 29 32	11 29 53	6	-11 5	6	0	-11 5	0 35	70 0	78 5
11 29 53	11 30 01	7	-12 0	3	0	-11 6	0 [illegible]	71 4	75 0
11 30 01	11 30 17	8	-12 0	4	0	-11 7	0 [illegible]	73 9	72 [illegible]
11 30 17	11 30 41	9	-9 5	9	0	-11 [illegible]	0 [illegible]	72 0	[illegible]
11 30 41	11 31 01	10	-10 0	4	0	-11 [illegible]	0 55	72 0	63 9
11 31 01	11 31 17	11	-10 0	4	0	-11 [illegible]	0 54	74 0	67 7
11 31 17	11 31 17	(END/BLOCK)		0	0				
11 32 01	11 33 32	(TRAINING)		26	0				
11 33 32	11 34 17	12	-10 9	13	0	11 [illegible]	0 50	69 4	71 4
11 34 17	11 34 42	13	-10 9	7	0	-11 1	0 46	70 0	71 [illegible]
11 34 42	11 35 06	14	-9 4	7	0	-11 0	0 48	67 4	71 4
11 35 06	11 35 24	15	-9 4	5	0	-10 9	0 49	68 1	71 7
11 35 24	11 35 34	16	-9 4	3	0	-10 [illegible]	0 50	67 3	72 3
11 35 34	11 36 02	17	-10 4	8	0	-10 [illegible]	0 47	67 3	75 0
11 36 02	11 36 19	18	-11 4	5	0	-10 [illegible]	0 45	67 3	75 4
11 36 19	11 36 31	19	-11 4	3	0	-10 9	0 43	67 3	75 4
11 36 31	11 37 01	20	-10 9	9	0	-10 9	0 40	67 7	76 2
11 37 01	11 37 01	(END/BLOCK)		0	0				

Figure A-2. Sample Line Printer Output.

TEST INSTRUCTIONS

This experiment will test your ability to hear a faint signal. The signal will always occur in distinct trials. Each trial is composed of two listening intervals. The buttons in front of you labelled "1" and "2" light up during the first and second listening intervals of each trial, respectively. After listening during the two intervals, your job is to push either Button "1" or "2", corresponding to the listening interval in which you think the signal has occurred.

The signal you are listening for will occur equally often in each listening interval, but in random sequence. Since there is no pattern whatsoever to the order in which the signal occurs in the two listening intervals, you must base your decision solely upon what you hear.

A constant background of noise will be playing in the chamber at all times. The signal will be easy to hear when the test first begins. The level of the signal will be gradually adjusted according to your correct or incorrect decisions. When you make a correct response, the level of the signal is lowered. When you make an incorrect response, the level of the signal is raised. On some trials the level of the signal will be so low that you will have difficulty deciding which listening interval contained the signal.

Regardless of how confident you feel about your decision, you must push one of the two buttons on each trial. You have at least a 50 percent chance of getting a correct answer just by guessing.

If you do not make a response, the level of the signal remains the same. This prolongs the length of the experiment considerably. Remember that the signal always occurs during either interval 1 or 2 of each trial, even if you are not certain of the interval in which it occurs.

You must respond in the one second interval following the presentation of the two listening intervals. Do not respond before you see the light corresponding to interval 2 go out. Directly after your response on each trial, the button corresponding to the listening interval in which the sound actually did occur will light up briefly. This will let you know immediately whether your decision was correct or incorrect.

An experimental session continues until your percentage of correct decisions has reached a predetermined level. In order to end an experimental session as rapidly as possible, it is therefore necessary for you to listen very carefully on each trial, and to make your decision as best you can.

APPENDIX B
GRAPHICAL SUMMARY OF
INDIVIDUAL SUBJECT DATA

This appendix contains plots of detection data for each test subject. Tabular data from these graphs may be found in Table 5 of this report.

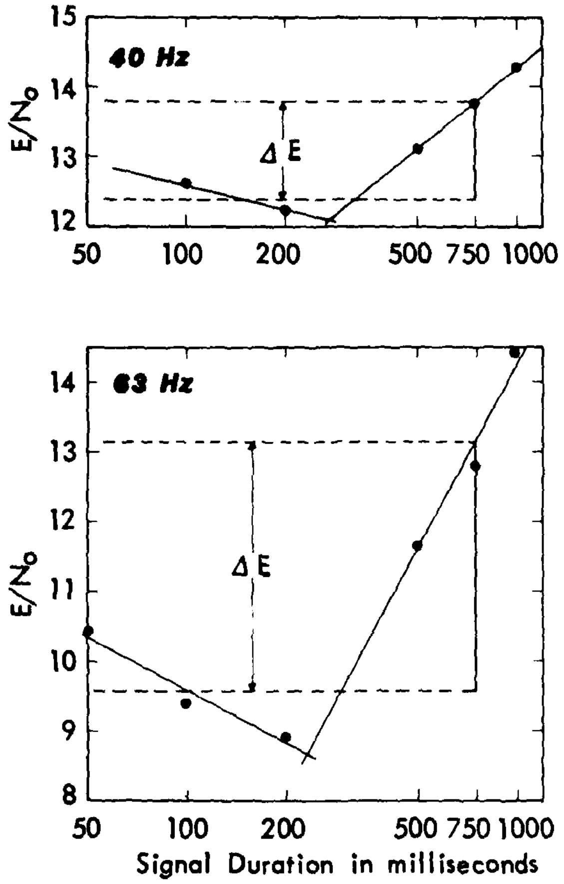

Figure B-1. E/N_o Vs Duration Curves for Subject 1.

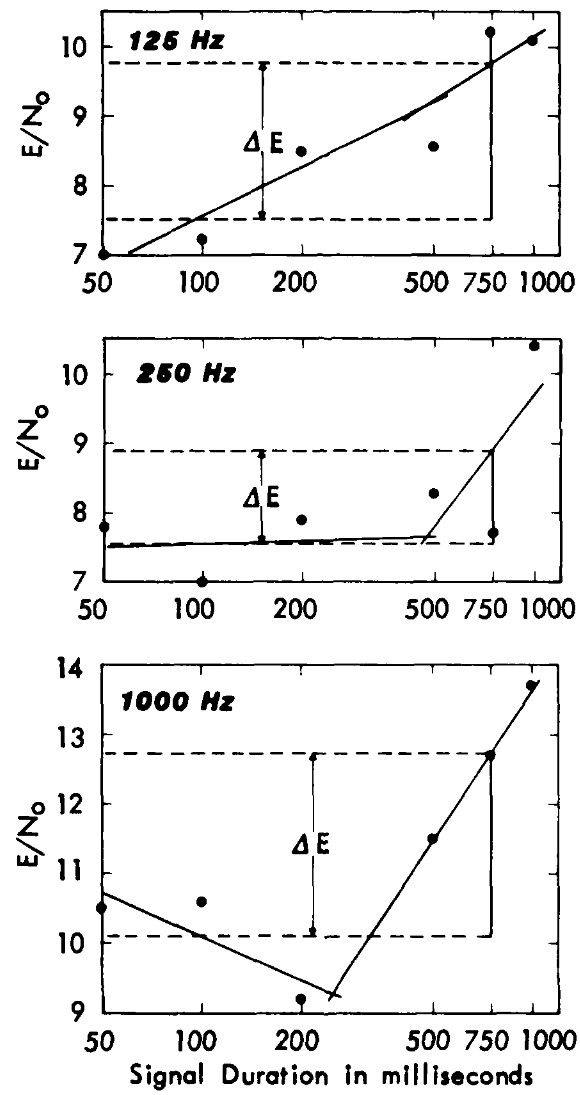

Figure B-1. Continued.

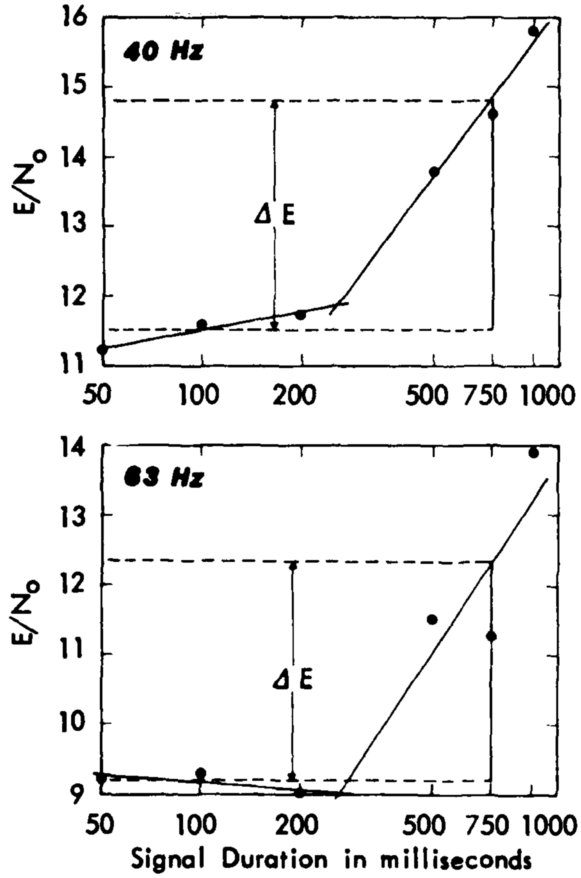

Figure B-2. E/N_o Vs Duration Curves for Subject 2.

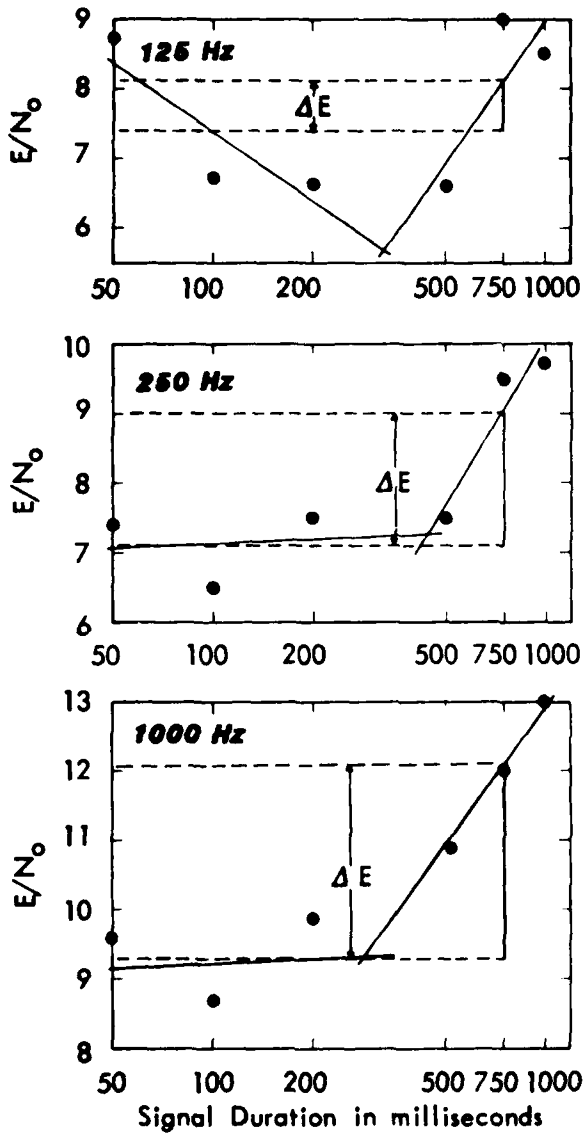

Figure B-2. Continued.

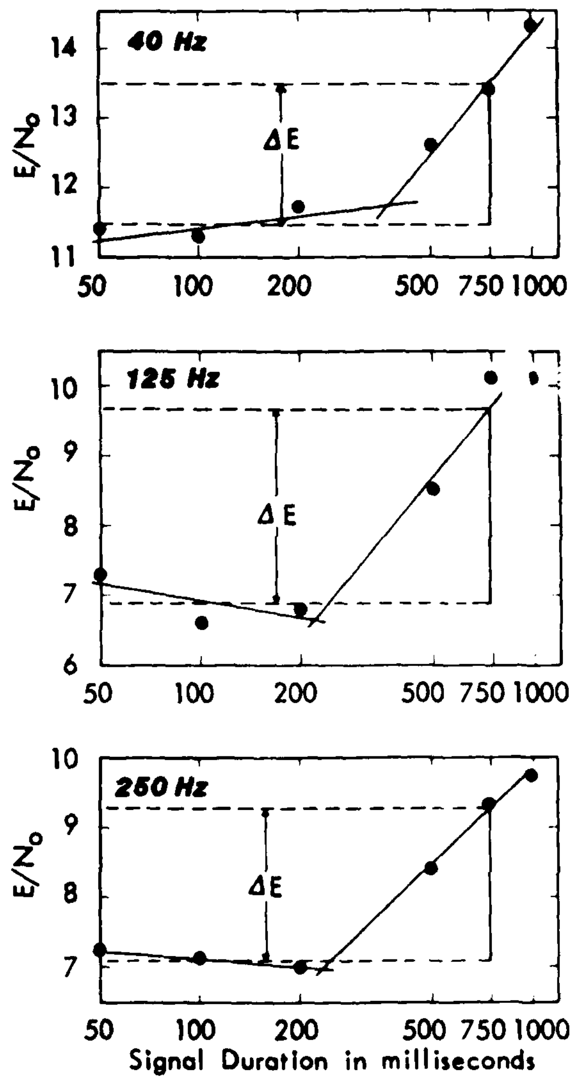

Figure B-3. E/N_o Vs Duration Curves for Subject 3.

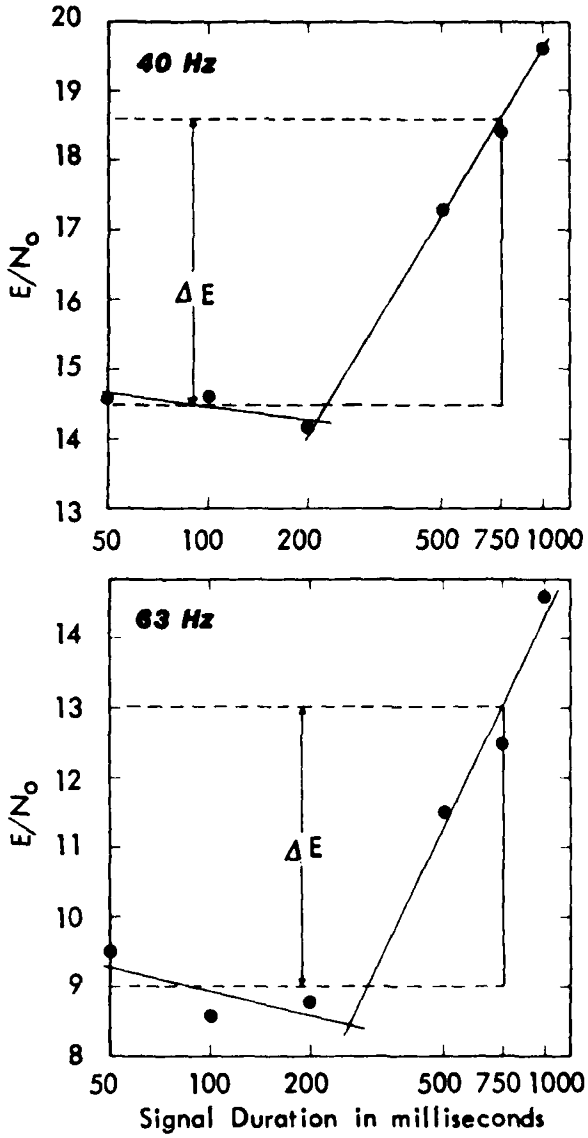

Figure B-4. E/N_o Vs Duration Curves for Subject 4.

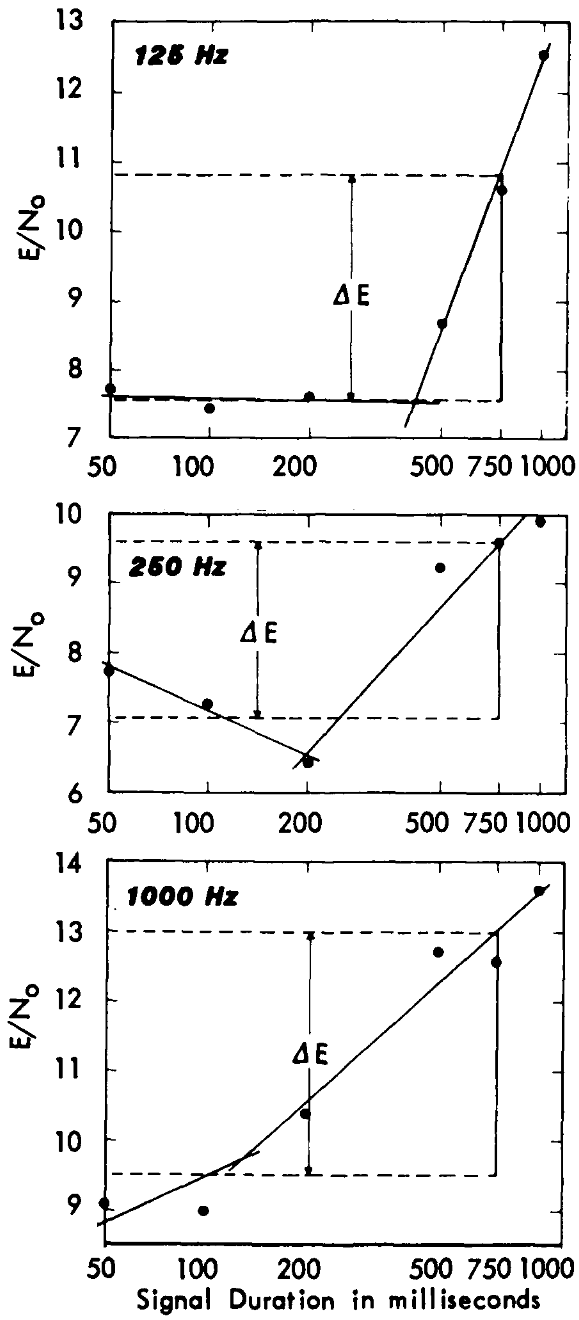

Figure B-4. Continued.

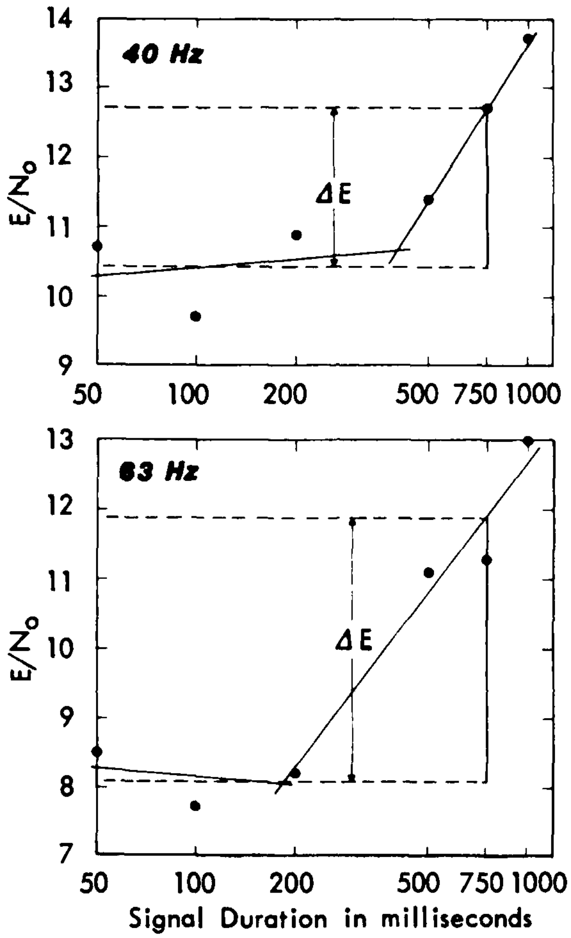

Figure B-5. E/N_o Vs Duration Curves for Subject 5.

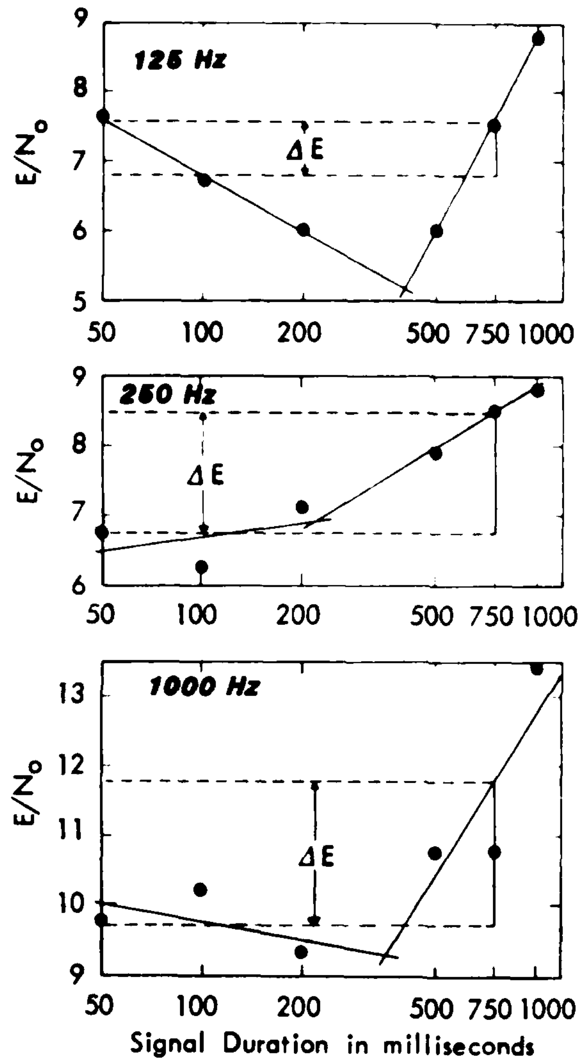

Figure B-5. Continued.

APPENDIX C
TERMINOLOGY FOR SIGNAL TO NOISE RATIO MEASUREMENTS

Throughout the text of this report, signal to noise ratio quantities are expressed in different forms (e.g., S/N, E/N_o, or in words) as appropriate to the context of the discussion. In all cases, the intent of the terminology is to express the division of a quantitative measure of signal intensity by a quantitative measure of noise intensity. The bandwidth of the signal measurement is implicitly taken to be 1 Hz, since the signals of interest are pure sinusoids. The bandwidth of the noise measurement is also 1 Hz, unless otherwise stated.

Consider, for example, the quotient of signal power (in watts) and noise power (also in watts). Dividing signal power by noise power yields the dimensionless ratio referred to as "signal to noise ratio". This ratio can also be expressed in decibels by multiplying the logarithm (base 10) of the quotient by 10, as:

$$\text{ratio in decibels} = 10 \log \left(\frac{S}{N}\right) = 10 \log (S) - 10 \log (N)$$

where: $10 \log (S)$ = signal level in decibels
$10 \log (N)$ = noise level in decibels.

From the above equation it is clear that the ratio <u>in decibels</u> may be obtained by simply subtracting the noise level (in decibels) from the signal level (in decibels). Since sound levels (signal or noise) are both conveniently measured in decibels, it is most convenient to express their power ratio in decibels as well. Unless otherwise stated, all signal to noise ratios in this report are presented in decibels, along with a brief description of how the signal and noise quantities were each measured.

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