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1974

COMPUTER GENERATED TONES IN VARIABLE BANDWIDTH NOISE AND ITS APPLICATION TO A STUDY OF SUB-CRITICAL DETECTION

by

MILTON ROSENSTEIN

A dissertation submitted to the Graduate Faculty in Speech in partial fulfillment of the requirements for the degree of Doctor of Philosophy, The City University of New York

1974

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Supervisory Committee

The City University of New York

Abstract

COMPUTER GENERATED TONES IN VARIABLE BANDWIDTH NOISE
AND ITS APPLICATION TO A STUDY OF SUB-CRITICAL
DETECTION

by

MILTON ROSENSTEIN

Advisor: Professor Harry Levitt

A computer program designed to minimize the limitations of hardware tone and noise sources and to provide precise control of tone and noise parameters, was developed and applied to the study of the detection of tones in variable bandwidth noise. In this study, which used two repeated (frozen) noises per bandwidth, the mean intensity of each noise presentation and the starting phase of the tone referred to the noise, was held constant and equal throughout the tests while the noise bandwidth ranged from 1.25 to 640 Hz. Despite these identical presentation-wide mean intensities, two threshold noise bandwidth curves emerged differing significantly in the sub-critical region; one curve fitted by a straight line satisfying the energy detection hypothesis, the other, fitted by a curve exhibiting a pronounced hump between noise bandwidths of 10 and 40 Hz. These results, which contradict the hypothesis holding that energy or intensity ratios between signal and noise, alone determine sub-critical thresholds, were linked to differences in two parameters of the noises; (1) the mean value

of the noise lying under the on-transition of the tone, and (2) signal-like excursions of the noise envelope in approximately the same region. These results suggested a hypothesis concerning sub-critical detection which provided a resolution of apparent contradictions re: sub-critical thresholds reported by some previous investigators.

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This work is dedicated to my wife, Rebecca, for her understanding, patience and constant encouragement and to my daughters Louise and Amy for their faith and love.

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INTRODUCTION

Psychoacoustics studies the dynamics of hearing. These studies are typically based upon the responses of human subjects to various acoustic stimuli. Though this procedure is inherently subjective, it is the goal of investigators in this area to secure results which are as objective as possible and which will be repeatable by others. This goal is made approachable by restricting the subject to simple responses such as "yes" or "no" - meaning I hear, or do not hear, the signal, or to the choice of one interval from a group of intervals, as the one containing the desired event.

By this severe limitation upon response categories, subject reliability and test objectivity are improved but the extensive ensemble of potential human reactions to acoustic events is rendered "off limits" to psychoacoustics. This limitation means that attempts to uncover the complex secrets of hearing must rely mainly upon the sophistication and precision of the test signals rather than upon the subtleties of responses. From this viewpoint, the task of the psychoacoustic investigator appear similar to that facing a questioner in a game of twenty questions played over a noisy telephone line with respondents who do not always understand the question.

Progress in psychoacoustics has usually been closely related to the state of the technology of signal generation and measurement. A contrasting of the efforts involved in early attempts to investigate

low frequency hearing using huge mechanical sources to the ease with which such experiments may now be performed, dramatizes this relationship, but even a cursory review of the widespread use of precise timing equipment, sophisticated active filters, digital noise sources, advanced spectral measuring devices, etc., confirms this dependency.

The most versatile and precise device, presently available, is the digital computer but though the computer has been widely used by investigators in psychoacoustics for analysis of results and, occasionally for signal generation, full utilization of its capabilities has not yet occurred. This study attempts to make a fuller utilization of these capabilities by establishing procedures for generating the test signals (tones and noise) entirely by computer and the necessary peripherals. The belief that these procedures would provide investigators in the psychoacoustic area with a versatility in signal generation and a precision of control over such signals, exceeding that presently existing, supplies the primary motivation for their development.

The dissertation consists of three sections. Section 1 (Chapter I) describes computer software which generates a useful class of test signals, the pure tone masked by noise of controllable bandwidth, and discusses the considerations underlying the development of such software. Section 2 (Chapters II, III and IV) first reviews the pertinent literature and then describes an application of the computer generation program to an investigation of pure tone detection in masking noise diminishing in bandwidth from supra-critical width to extremely narrow bands; a topic of interest to psychoacousticians and one containing unresolved problems and

contradictions as well. Since investigators in this area have, typically, been handicapped by the severe technical difficulties associated with the control of hardware noise and the implementation of the required very narrow band filters; this seemed a particularly apt application. In addition, Section 2 served also to evaluate the computer generation program and its supporting software. This evaluation is discussed in the concluding sections of Chapter 1 and further, in Section 3 (Chapter V). Modifications of the program suggested by this evaluation are also discussed in this chapter.

CHAPTER 1

COMPUTER GENERATION OF TONES IN ADDITIVE NOISE OF CONTROLLABLE BANDWIDTH

I. Motivation

The motivation for the development of a computer program to generate pure tones and noise for psychoacoustic research arises primarily from restrictions placed on such research by the difficulties associated with specifying and analyzing hardware generated noise and the practical limitations placed upon filter bandwidths, shapes precision, and transient response by the use of analogue components. The first restriction has enforced a statistical view of noise and a consequent emphasis upon long term statistical measures as the major factors of random stimuli. The second restriction has hampered research in those areas where extremely narrow bandwidth noise has been required, or has led to the acceptance of doubtful assumptions about the nature of noise in order to evade these difficulties. The use of a computer to generate this class of signals can, on the basis of previous experience with digital generation techniques, be expected to overcome these restrictions.

II. Noise

A. Hardware Generated Noise

Aside from the use of deterministic signals such as tones, pulses, clicks, etc., random noise is typically used in psycho-

acoustic investigations to mask the detection of signals. For the purposes of such research, this noise may be defined as a signal with varied degrees of unpredictability. Unpredictability reflects an inability on the part of the hearer to anticipate succeeding signal parameters on the basis of those preceding; the degree of unpredictability is quantifiable in terms of auto-correlation.

The inherent unpredictability of hardware-generated noise has restricted its specification, typically, to measures based upon statistical averages occuring over long periods of time. These measures are illustrated by the shape of the probability density distribution, its mean value, its standard deviation, and the maximun rate at which the noise envelope changes. A typical set of specifications might include:

1. Probability density distribution...Gaussian
2. Mean or DC value.....0 volts
3. Standard deviation (RMS).....10 volts
4. Bandwidth.....100 Hz

Measures "1", "2", and "3", as is well known, are characteristics of the probability density distribution. Measure "4", however is independent of this curve so that changing the noise bandwidth does not affect the probability density distribution.

These measures, are exact only for a noise sample of very long duration or as the average of many shorter samples. A single 400 msec. noise burst extracted from a 5 minute sample with the above specifications, is likely to have very different characteristics than those specified as long term statistical averages. The investigator justifies his use of these long term specifications on the

basis that the mean values of the many brief samples presented to the subject approach the statistical specifications. While there is no doubt that this is valid statistically, it is not established, however, that this is true in a given psychoacoustic experiment in which subject response variability interacts with noise sample variability.

B. Band-Limited Noise

Band-limited noise has had widespread application in investigations involving masking, particularly those concerned with detection within and near the borders of the critical band. Since other parameters such as the center frequency of the noise, its probability density distribution, and intensity, are independent of the bandwidth, varying the bandwidth introduces an additional orthogonal dimension to the test ensemble, thereby increasing test sophistication.

This additional dimension has proven to be a fruitful one since a number of characteristics of hearing seem to be affected by bandwidth; for example, the threshold of a pure tone masked by noise (Fletcher, 1940; Schafer, Gales, Shewmaker, and Thompson, 1950), the loudness of a constant intensity noise (Zwicker, Flottorp, and Stevens, 1957), and the degree of transient masking of a signal immersed in noise (Zwicker, 1965). The fertility of the noise bandwidth parameter, it will be shown, may be due to the complex and changing array of envelope configurations presented to the hearer as bandwidth is changed evoking, apparently, revealing and sophisticated reactions.

C. Computer Generated Noise

Unpredictability

When noise is generated by computer the concept of unpredictability used above must be limited to unpredictability to the subject since computer generated noise, being recreatable, is completely predictable to any agent which either knows or can deduce the noise algorithm and the starting sequence. Nevertheless, computer noise, carefully formulated, can satisfy the essential requirements of psychoacoustic noise by remaining unpredictable to the subject to the degree desired. There is, however, a gray area for cases in which the noise delivered to the subject is repeated for significant periods of time. In this case, the signal may be reacted to by the subject, either as a true noise or as a partially, or completely, predictable pattern depending upon his ability to memorize the signal.

Advantages

Specification - Because computer noise is completely specified by its generating software, it is not necessary to restrict parameter specification to long term statistical measures. The range of potential parameter specification runs from the traditional long term measures such as overall mean intensity, bandwidth, and probability density distribution through similar short term specifications for individual presentations, to the specification and control of specific features of the individual stimuli. In addition, the independent test variable need not be restricted to those parameters, such as noise bandwidth or signal to noise ratio, which are most conveniently manipulated by hardware sources. Instead numerous other measures

are available as for example, grouping on the basis of measures of correlation, peak to valley ratios, local intensity variations, or complexity of patterning. As a result the investigator is provided with a potentially interesting set of parameters which, in addition to average measures, may be operative factors in hearing.

Another less obvious result of the extension of versatility in parameter control is the weakening of the dependence upon hardware dictated parameters and the tendency to accept these parameters as fundamental factors in hearing. To a discipline so dependent upon the scope of its "questions," it is important not to be restricted by equipment considerations. Computer generation encourages the search for, and the exploitation of, such possibilities.

Narrow Bands - The practical difficulties which place a lower limit on the bandwidth of analog filters, arise from the nature of the components used to implement these filters, namely resistors, inductors, capacitors, and, for active filters, transistors. Unfortunately these elements do not supply just resistance, inductance, capacitance, and amplification. Each resistor, unavoidably, exhibits capacitance and inductance as well; each capacitor, resistance and inductance; each inductance, resistance and capacitance; and each transistor, resistance, inductance, and capacitance. In addition, all these characteristics vary quantitatively with temperature, age, sometimes humidity, and, in the case of active devices, voltage and current as well. As a result though theoretical designs exist for filters with any center frequency to bandwidth ratio, the required components are unobtainable. This practical difficulty has either

set a lower limit on the narrowest bandwidths used in investigations or has inspired doubtful strategems to bypass this impediment.

Computer simulation of analog filtering provides one escape from these difficulties. The computer, operating as it does, with numeric streams rather than real components, encounters none of the difficulties discussed above. Consequently the achievable filter specifications are limited only by the precision of the computations and the ingenuity of the programmer. Simulated filters have, of course, their own problems, namely that of errors introduced by the need to round off, or truncate numbers, but at present these limitations are far less restrictive than those applying to hardware filters. The computer generation program demonstrates this capability, permitting the simulation of filters with center frequency to bandwidth ratios beyond the capacity of analog filters.

Capabilities - In addition to generating conventional types of noise processed through simulated filters, the computer program offers a potential for the generation of novel signals not feasible with hardware sources. To generate tone in noise signals, the used is required to specify items such as signal frequency, center frequency of the noise, filter shape and bandwidth, etc. (see Table 1). For the generation of conventional signals, some of these parameters remain fixed for each run, others for each trial, and the remaining parameters, for each presentation. However the program offers the prospect of changing some or all of these parameters during the course of an individual presentation in accordance with specific strategems.

This is an engaging capability and suggests numerous extensions beyond conventional test signals. A few examples of signals which

TABLE 1

Direct Specifications

Type	Range	Comments
Noise		
Center Frequency	0 to 5000 Hz.	Limited by sampling rate
Bandwidth	.001 to 5000 Hz	Limited by sampling rate
Selectivity	First to tenth order	Limited only by logistics
Rms	Computer limited	.....
Probability Densities	Uniform, Rayleigh, Gaussian	Can be extended
Pure Tone		
Frequency	0 to 5000	.....
Both		
Rise and Fall Shapes	Step, Expo., Sine, Ramp	.....
Rise and Fall Times	0 to duration limits	Measured to 90% max
Duration	Unlimited	Except by logistics
Onset--Tone re: Noise	Unlimited	Positive or Negative Delays
Pure Tone to Noise Ratio	-400 to 20 dB	Typical not limit

may be generated by the computer program but which are not feasible for hardware noise sources include presentations with changing noise bandwidths, changing center frequencies, changing tone frequencies, varying intensity concentrations, etc. The list is open ended and reflects the versatility in signal generation arising from control of the micro-structure of the stimulus.

Other Advantages - The use of the computer generation program offers two additional advantages. First, model formulation and testing is epedited because both stimuli and responses lie in computer memory. This permits the analytical model to react to the same stimuli as did the human subjects. In addition since the original stimuli is available for repeated detailed analysis, the model may be formulated in terms of the microstructure of the stimuli rather that upon its global averages. Finally the huge data processing capability of the computer supplies the individual with the ability to carry out large scale projects formerly beyond his scope.

III. Signal Generation Program - General

A. Program Form

The signal generation program is in the form called "next state." This means that a single pass through the program is required to generate one word of output. To generate additional output words, the system control is returned to the beginning of the active program. If 4000 output values are required, for example, then 4000 passes would be needed. Each stage of the program generates the few intermediate values required for the following stage

and then clears its variables. The final output is written into peripheral storage and all variables are available for the next pass. The converse of the next state form is one in which all intermediate outputs are calculated and stored in a single pass and are passed, in their entirety, to the next stage of calculations. In this latter case, the run is completed in a single pass.

The major advantage of the next state form lies in its ability to minimize memory requirements. Since one of the goals set for the generation program is the capability to produce signals of very long duration, this feature is important. Signals generated by next state programs are limited only by the availability of computer time and the extent of peripheral storage.

One disadvantage of next state programs arises from the need to continually input and output data from or into peripheral storage. Computers can access information from peripheral storage much more quickly when such information is transferred at one calling rather than by repeated access. This difference is far from theoretical and necessitated modification of the next state form for practical reasons.

B. A Simplified Flow Chart

Figure 1 is the flow chart of a simplified version of the generation program which will serve to present the underlying concepts. Subsequently, sections of the program will be discussed in greater detail.

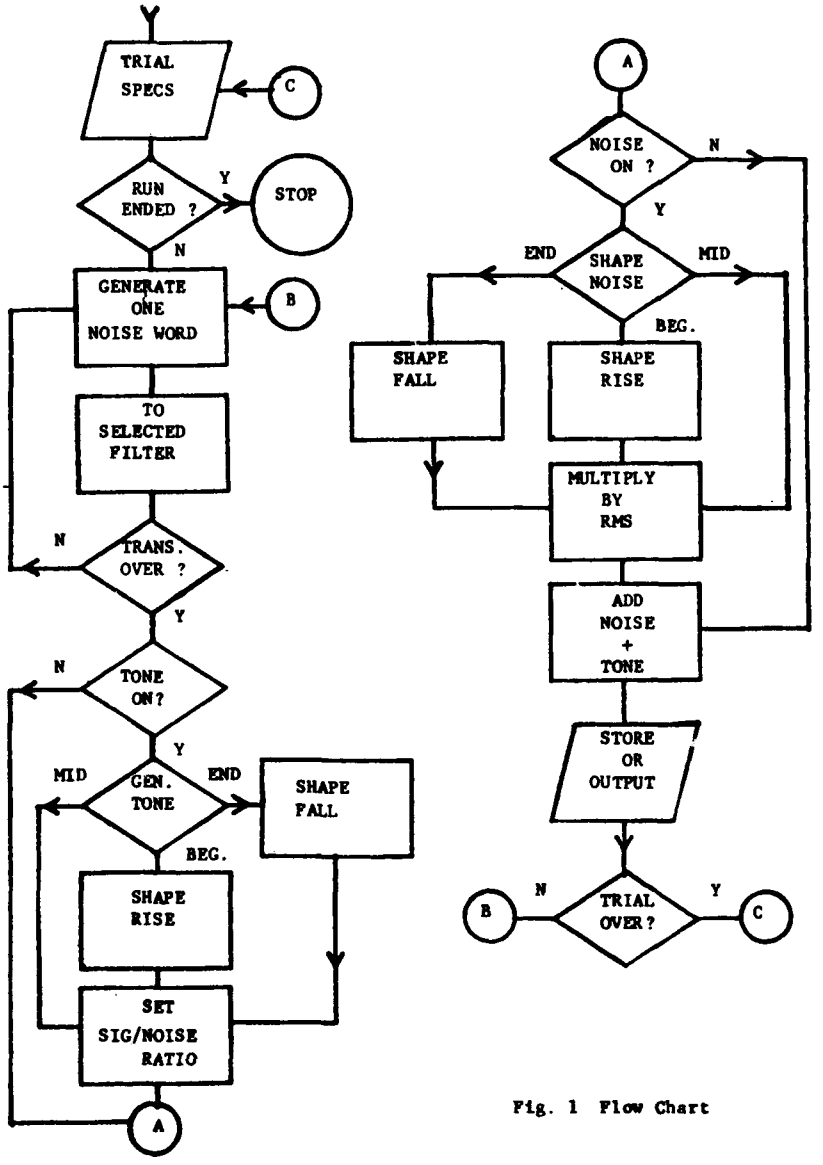


Fig. 1 Flow Chart

The box labelled "Trial Specs" represents the section of the program in which specifications for the pure tone and noise, are received. The specifications required by the program are given in Table 1.. Some of these specifications may be fixed for the entire test and so will be read in only once, while others may be changed after each run, or after each presentation. (It is also possible to change specifications after each pass). Parameters which are changed rapidly are supplied by an auxilliary program which prepares an appropriate input tape.

The box labelled "Run Ended" represents the program section which governs the extent of a run. Since the investigator is expected to divide the total test into runs, provision is made to terminate the program execution after the desired number of trials and to resume at the consecutive one.

The box labelled "Generate One Noise Word" represents the program section which generates one output value of the broad band noise which subsequently feeds the filters.

The box labelled "To Selected Filter" and the one following, "Trans. Over", represents the section which directs the broad band noise word to the selected filter but blocks the filter output until initial transients have subsided. This action permits envelope shaping to take place without regard to the transient response of the particular filter.

The boxes labelled "Tone On ?" and "Noise On ?" represent the program section which control the initiation of the tone or noise signals. The two decision boxes act to switch on the

tone or noise according to delays selected. The tone is turned on by initiating its generation; the noise, however, is always available after the transient suppression time has elapsed but until wanted, it is suppressed by setting its magnitude at zero.

The boxes labelled "Gen. Tone," "Shape Fall," "Shape Rise," "Set Sig/Noise Ratio," "Shape Noise," and "Multiply by RMS" represent the program's mechanism for shaping the envelope of the tone and noise separately. In particular, the "Set Sig/Noise Ratio" and the "Multiply by RMS" boxes represent the sections which adjust the relative amplitudes of the tone and noise. The RMS of the noise is set in relation to a calibrating tone generated at the start of each run; the tone level is derived from the signal to noise specification given in dB.

The boxes labelled "Add Noise + Tone" and "Store or Output" represent the sections of the program which combine the tone and noise and either stores the result on a peripheral storage device, or feeds it directly to a digital-analog converter for immediate conversion to audio.

The steps described above, constitute a single pass through the system and generate a single output value. The box labelled "Trial Over" returns control to "Generate One Noise Word" for the next pass until a trial or presentation has been completed. When this occurs, control is returned to the "Trial Specs" box where pertinent specifications are changed and a new trial or presentation begins.

The simplified program described above illustrates the single pass mode of the generation program. In this mode, the entire set of calculations is performed each time signals are generated. This mode is necessary for the production of non-repetitive noise but is inefficient for repetitive noise. In this latter case where the same noise is reused a significant proportion of the time and where the entire ensemble of noises is not very large, the double pass mode may be more efficient. In this mode, the ensemble of noises is generated in one run and recorded in digital form. When signals are being generated, the appropriate noise samples are retrieved as needed, modified as necessary, and combined with tones. This action replaces extensive calculations required for the original generation of the noise.

This double pass mode permits the building of an ensemble of noises with novel characteristics not accessible to direct specification and beyond the capability of non-computerized systems. Noise ensembles characterized by factors related to correlation, to specific distributions of energy, to patterning, or to factors yet unknown, may be established by a double pass, generate, test, store, or reject mode. In the first passes, noises specified by a related, direct specification, are generated, tested for the indirect specification, and stored if acceptable. The noise ensemble assembled by repeated passes

is used in the coventional double pass mode for the generation of signals for succeeding runs.

C. Input and Output Considerations

The need to input voluminous specifications and to receive and analyze extensive arrays of subject responses, create the requirement for a number of supporting programs in order to fully realize the practical advantages of the generation program. In addition the large number of words output by the generation program raises logistical considerations of practical importance.

Typically, the specifications for a run will include fixed and variable ones, The variable ones will, most likely, be changed on a per presentation basis and randomized according to some strategy. It would be inefficient not to use the computer capabilities for this task. Two programs for the preparation of variable specifications and for encoding them onto data files, have been prepared and used in the investigation of Section 2. Each program uses a different strategy for the preparation and randomization of specifications. Such a program is probably required for every full scale investigation but its form will depend upon the specific investigation.

For most efficient use of the computer, the subject's responses should reside in memory along with the battery of generation and supporting programs. This not only facili-

tates sophisticated analysis but also simplifies and ensures accuracy in the checking of results. The subject's responses are typically "yes" or "no" so that entry by hand into a data file is not impractical but it is inefficient and increases the chances of error. Three schemes suggest themselves. One involves preparing sets of punched cards bearing required identifications and prepunched with either a "1" or "0". During the test the subject selects the desired card. If this physical activity is considered undesirable, then a more elaborate system, in which the subject's activation of a yes-no switch punches a record on computer readable media, may be used. Finally, if a dedicated computer is available, the results may be entered directly into computer memory. This last possibility presents many attractive potentialities such as adaptive programming, and immediate analysis and display of results, for example.

The logistics of storage for the output signals of the generation program create practical problems. For example, the investigation of Section 2 requires twenty runs, each run containing 320 presentations plus intervening pauses; each presentation composed of four, 1000 word records. If the intervening pauses are stored along with the signals, on digital tape, the amount of tape required becomes large. While this procedure is feasible, a more efficient use of digital tape is made by packing the signals together during

generation and introducing the pauses during readout. Noise and tone signals may also be generated and stored on separate channels of audio tape. This procedure reduces the output requirements to a minimum by permitting the system to function as a precise audiometer. The generation program possesses this capability requiring only slight modification of its output.

IV. Signal Generation - Specific

A. Introduction

The generation program requires algorithms for: (1) the production of uniformly distributed random numbers, (2) the conversion of this distribution to a Rayleigh distribution, (3) the conversion of the Rayleigh distribution to Gaussian, (4) the generation of a long duration sinusoid, (5) the simulation of a single tuned band pass filter, and (6) the combination of single tuned filters into higher order Butterworth filters. It is necessary to examine these algorithms in detail.

B. Uniformly Distributed Random Numbers

The algorithm used here is the additive process

$$X(J) = (X(J) + X(J-N)) \quad \text{MODULUS } 1.$$

Starting with an initial set of "N" uniformly distributed

random numbers in the interval $0 \leq X < 1$, the process generates the additional number, X(J), within the same interval. The sequence of numbers so produced, exhibits many, but not all, the characteristics of a pure, uniformly distributed, low correlation, random sequence which consequently is called "pseudo random." This distinction is important and will be discussed later.

The process generates numbers by adding the first and last numbers of a progressively incrementing sequence of "N" numbers followed by the modulus operation. Ideally any number of the sequence may appear with equal probability and complete independence in the X(J-1) or X (J-N) position. Adding these two numbers together, produces a probability density distribution, or histogram, given by the convolution of two equal uniform distributions and is given by

$$Y(NT) = \sum_{M=0}^{N} K_1(MT) \quad K_1(NT-MT)$$

where K_1 (MT) is a rectangular (uniform) distribution and K_1(NT-MT) is the same distribution reversed and incrementally delayed. (Digital convolution is discussed by Gold and Rader, 1969). The result of this convolution is a triangular distribution with a range from "0" to just below "2" and a maximum at "1." Imposing modulus "1" on this result, subtracts "1" from numbers equal to or exceeding "1" and trans-

poses the upper half of the curve into the "0" to just below "1" range. Algebraic addition of probabilities yields the desired uniform distribution.

This theoretical result was tested empirically for the algorithm used in the investigation of Section 2 of this dissertation in which "N" equals 50. The computer printout, Figure 2, shows the histogram for 10,000 numbers. The tendency towards a uniform distribution is evident. Figure 3, which is the curve of Figure 2, integrated, shows the summed probability for random numbers from "0" to the abscissa value. The expected result, a ramp saturating at 10,000 is closely approximated.

Green, Smith, and Klem (1959) tested the binary form of the algorithm

$$X(J) = (X(J-1) + X(J-N)) \text{ MODULUS } 2^n$$

for 10,000 samples with various "N"s, N=16, and N=32, and for numbers with 15 and 35 significant bits. The results were separated into octal digits starting at the decimal point and a tally was made of the frequency of occurence of each octal numeral: for "R" (the number of significant bits) equal to 15 and for "N" equal to 32, the first four octal bits supported the null hypothesis of pure uniformly random distribution but the fifth did not. For "R" equal 35, however,

the first ten digits supported the null hypothesis. While most large computers use word lengths of 32 bits corresponding to "R" = 32, smaller computers may be limited to 12 or 24 bits. It is sometimes necessary, therefore, to give some consideration to these matters. Since the requirements of the investigation of Section 2 imposed a 64 bit precision, the assumption of randomness seemed warranted for this application of the generation program.

Correlation and Periodicity

Another requirement of random number sequences is a low serial correlation between generated numbers or groups of numbers. Ideally this implies statistical independence and the absence of patterning. Since computer generated pseudo random numbers may exhibit periodicity and patterning, this characteristic requires investigation. Assuming that the correlations would persist through transformations of the probability density distribution from uniform through Rayleigh to Gaussian, the test for serial correlation was carried out on the Gaussian distribution.

FIGURE 2: Histogram of Uniformly Distributed Noise

The abscissa values given under "interval" bracket the interval within the noise values are summed. The ordinate values given under "no. of samples" give the sum plotted across from the upper end of the interval.

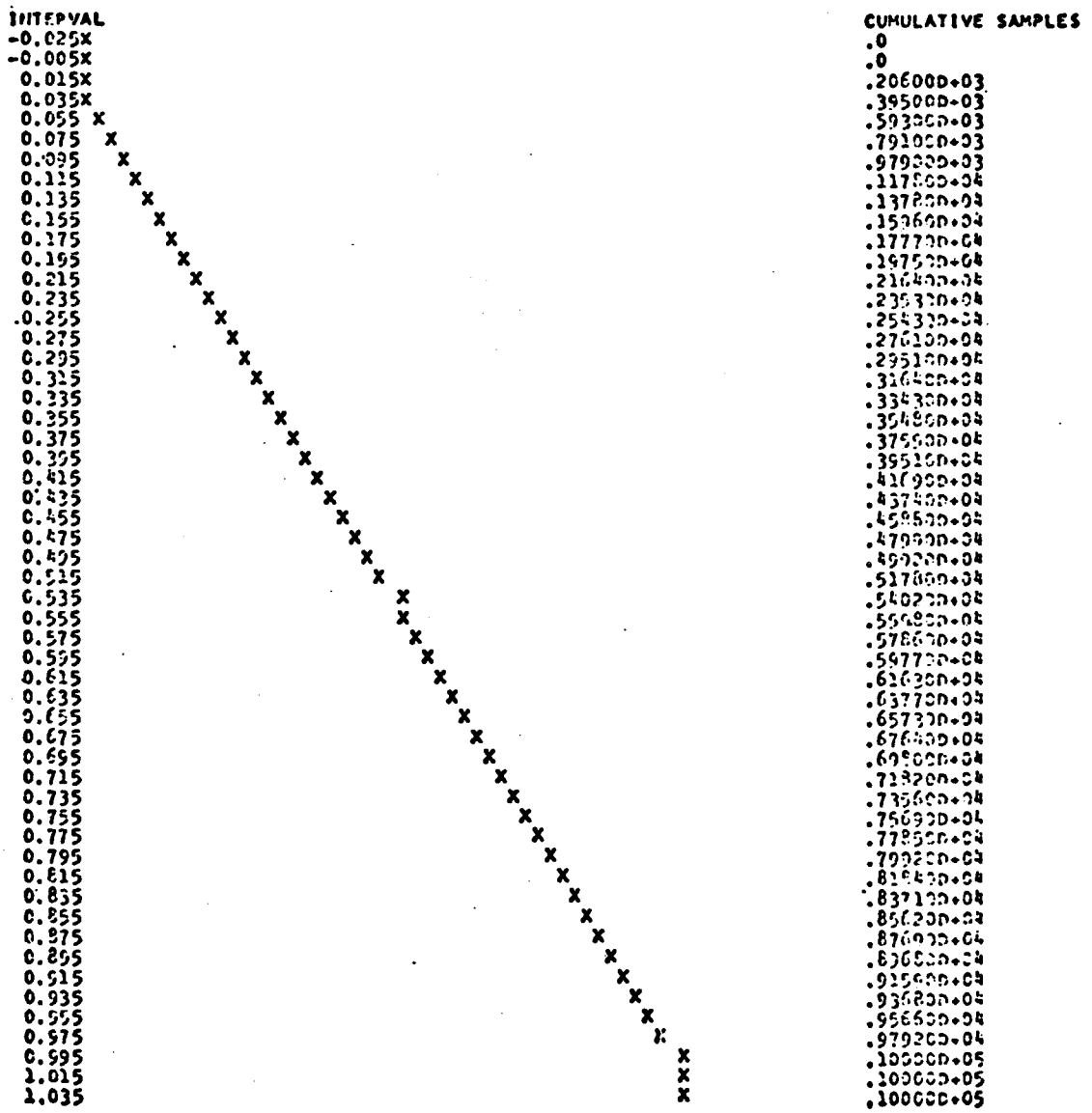
HISTOGRAM OF UNIFORMLY DISTRIBUTED NOISE

INTERVAL		NO. OF SAMPLES
-0.050X		.0
-0.030X		.0
-0.010X		.0
0.010		.20700D+03
0.030		.18900D+03
0.050		.19700D+03
0.070		.19800D+03
0.090		.18900D+03
0.110		.20000D+03
0.130		.20000D+03
0.150		.21700D+03
0.170		.17800D+03
0.190		.19600D+03
0.210		.18900D+03
0.230		.18900D+03
0.250		.19100D+03
0.270		.21800D+03
0.290		.19100D+03
0.310		.21500D+03
0.330		.17900D+03
0.350		.20600D+03
0.370		.21000D+03
0.390		.19100D+03
0.410		.21900D+03
0.430		.20800D+03
0.450		.21400D+03
0.470		.21200D+03
0.490		.19300D+03
0.510		.19600D+03
0.530		.22300D+03
0.550		.19500D+03
0.570		.18800D+03
0.590		.19100D+03
0.610		.18600D+03
0.630		.21600D+03
0.650		.19200D+03
0.670		.19200D+03
0.690		.21600D+03
0.710		.20200D+03
0.730		.17700D+03
0.750		.21300D+03
0.770		.21500D+03
0.790		.20700D+03
0.810		.19100D+03
0.830		.18600D+03
0.850		.19300D+03
0.870		.20600D+03
0.890		.20200D+03
0.910		.19200D+03
0.930		.20700D+03
0.950		.19600D+03
0.970		.22500D+03
0.990		.20800D+03
1.010X		.0
1.030X		.0

FIGURE 3: Integrated Histogram of Uniform Noise

The abscissa valuse given under "interval" represent the upper value to which the noise is summed. The lower end of the interval is always minus infinity. The ordinate values given under "cumulative samples" is the integrated sum.

INTEGRATED HISTOGRAM OF UNIFORM NOISE



A search of the auto-correlation function for peaks provide a sensitive test for correlation, periodicity, and patterning. The auto-correlation function is given by

$$\sum_{t=0}^{m} \sum_{i=1}^{n} X(i) \quad X(i+t)$$

where X(i) and X(i+t) are random numbers separated by "t" cycles, "n" is the total range over which the inner summation occurs, and "m" is the range of the separation variable "t." Figures 4 and 5 show a plot of the auto-correlation function with the maximum value normalized to "1," versus the separation "t." Each point is found by finding the product of all pairs of random numbers separated by the number of cycles shown on the abscissa and summing these pairs across the entire sample. For the test of 10,000 numbers, "n" varied from 1 to 6000 and "t" varied from 0 to 4000 so that the sample boundaries were never passed. Any patterning or serial correlation would appear as large peaks in the plot. Figure 5 is a magnified view of Figure 4 and both go only to an upper "t" of 55. The test, however, went to a

"t" of 4000 without finding a normalized auto-correlation value equal to or greater than 0.05. This indicates that samples up to 400 milliseconds at least, may be considered to be uncorrelated.

Green, Smith, and Klem (1969) support this conclusion but their "t" does not exceed 20. In addition, they investigated some other aspects of independence for the binary form of the algorithm. They investigated the mutual independence of various octal groups in different positions and the distribution of equal groups of numbers lying above and below an arbitrary constant. In both cases, they found the distributions to be both uniformly random and independent.

Departure From Randomness

Random numbers generated by algorithm, do not always obey the laws of probability. In a true uniform, uncorrelated distribution in which every number has a probability "p", a group of "n"

FIGURE 4 Plot of Autocorrelation Function - 1

The abscissa values given under "interval" give the separation "t" between the noise values. The ordinate values given under "correlation" are the values of the function versus "t".

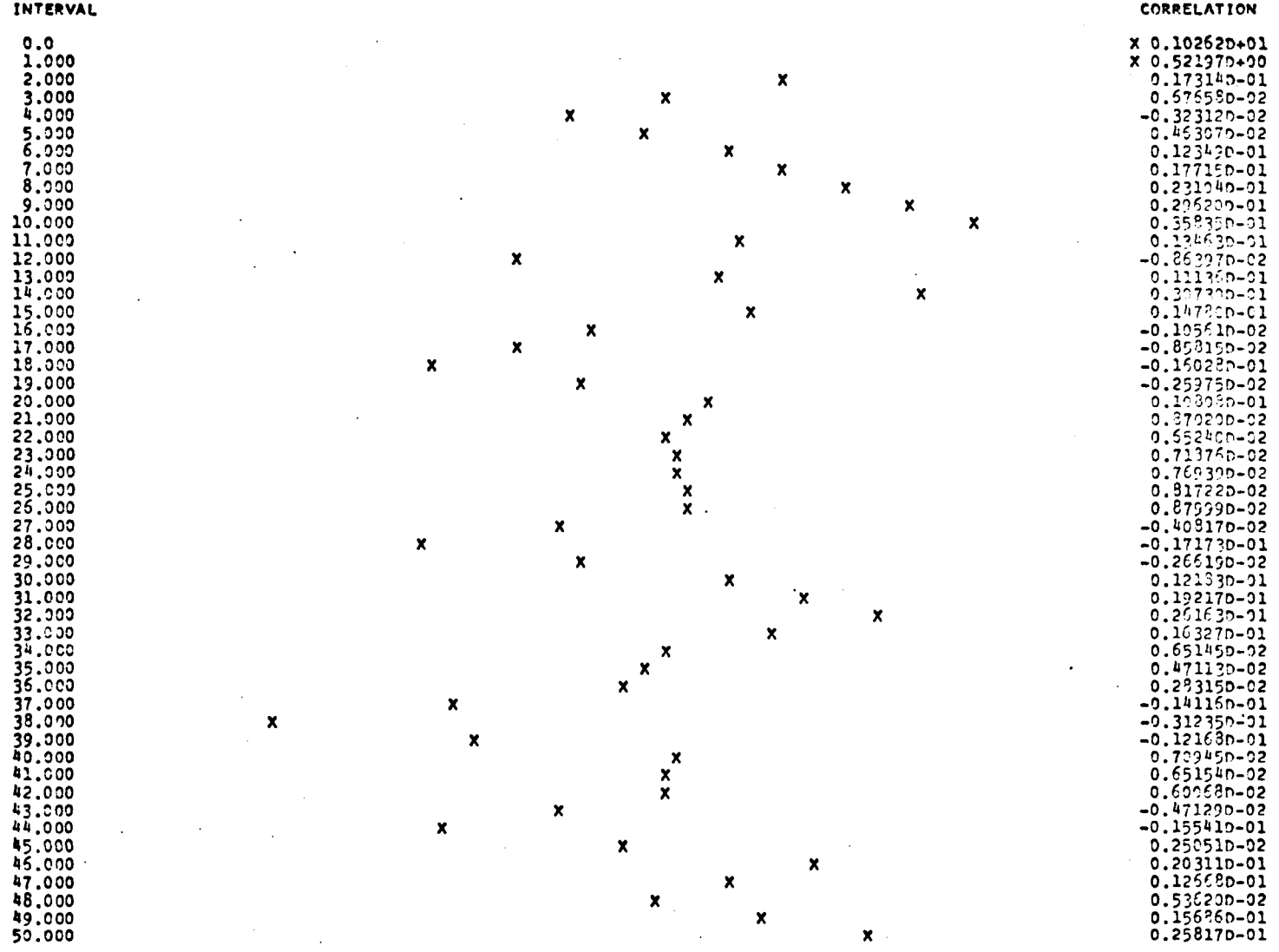
PLOT OF AUTOCORRELATION FUNCTION

INTERVAL		CORRELATION
0.0	x	0.10262D+01
1.000	x	0.52197D+00
2.000	x	0.17314D-01
3.000	x	0.67658D-02
4.000	x	-0.32312D-02
5.000	x	0.46307D-02
6.000	x	0.12349D-01
7.000	x	0.17715D-01
8.000	x	0.23104D-01
9.000	x	0.29620D-01
10.000	x	0.35835D-01
11.000	x	0.13463D-01
12.000	x	-0.86397D-02
13.000	x	0.11136D-01
14.000	x	0.30730D-01
15.000	x	0.14780D-01
16.000	x	-0.10561D-02
17.000	x	-0.85815D-02
18.000	x	-0.16028D-01
19.000	x	-0.25975D-02
20.000	x	0.10809D-01
21.000	x	0.87020D-02
22.000	x	0.65240D-02
23.000	x	0.71876D-02
24.000	x	0.75939D-02
25.000	x	0.81722D-02
26.000	x	0.87999D-02
27.000	x	-0.40817D-02
28.000	x	-0.17173D-01
29.000	x	-0.26619D-02
30.000	x	0.12183D-01
31.000	x	0.19217D-01
32.000	x	0.26163D-01
33.000	x	0.16327D-01
34.000	x	0.65145D-02
35.000	x	0.47113D-02
36.000	x	0.28315D-02
37.000	x	-0.14115D-01
38.000	x	-0.31235D-01
39.000	x	-0.12163D-01
40.000	x	0.70945D-02
41.000	x	0.65154D-02
42.000	x	0.60068D-02
43.000	x	-0.47129D-02
44.000	x	-0.15541D-01
45.000	x	0.25051D-02
46.000	x	0.20311D-01
47.000	x	0.12668D-01
48.000	x	0.53620D-02
49.000	x	0.15686D-01
50.000	x	0.25817D-01

FIGURE 5 Plot of Autocorrelation Function - 2

The abscissa values given under "interval" give the separation "t" between the noise values. The ordinate values given under "correlation" are the values of the function versus "t". Figure 5 is a magnified version of Figure 4.

PLOT OF AUTOCORRELATION FUNCTION



INTERVAL	CORRELATION
0.0	X 0.10262D+01
1.000	X 0.52197D+00
2.000	0.17314D-01
3.000	0.57659D-02
4.000	-0.32312D-02
5.000	0.46307D-02
6.000	0.12349D-01
7.000	0.17715D-01
8.000	0.23104D-01
9.000	0.29620D-01
10.000	0.35835D-01
11.000	0.13466D-01
12.000	-0.86397D-02
13.000	0.11136D-01
14.000	0.30739D-01
15.000	0.14728D-01
16.000	-0.10561D-02
17.000	-0.85815D-02
18.000	-0.16028D-01
19.000	-0.25975D-02
20.000	0.10309D-01
21.000	0.87020D-02
22.000	0.55240D-02
23.000	0.71376D-02
24.000	0.76939D-02
25.000	0.91722D-02
26.000	0.87999D-02
27.000	-0.40817D-02
28.000	-0.17173D-01
29.000	-0.26619D-02
30.000	0.12183D-01
31.000	0.19217D-01
32.000	0.26163D-01
33.000	0.16327D-01
34.000	0.65145D-02
35.000	0.47113D-02
36.000	0.28315D-02
37.000	-0.14116D-01
38.000	-0.31235D-01
39.000	-0.12168D-01
40.000	0.70945D-02
41.000	0.65154D-02
42.000	0.60068D-02
43.000	-0.47129D-02
44.000	-0.15541D-01
45.000	0.25051D-02
46.000	0.20311D-01
47.000	0.12568D-01
48.000	0.53620D-02
49.000	0.15626D-01
50.000	0.25817D-01

numbers has the probability of occurrence "p^n." Once a group of "n" has appeared its probability of recurrence remains "p^n." It therefore, may reappear and not introduce periodicity. The proceeding numbers are still random and non-periodic. In the case of the additive algorithm used here which has a starting group of "n" numbers, any recurrence of this group of numbers leads to the entire sequence of numbers being repeated cyclically. If periodicity is to be avoided the probability of the recurrence of any of the "n" groups must be made small by making "n" large. Therefore unlike the truly random case which involves selection with replacement, the pseudo-random case involves selection without replacement for groups of size "n" for non-periodic noise. Fundamentally the noise algorithm is a complex deterministic process which generates streams of uniformly distributed, uncorrelated numbers which individually follow the laws for uniform random distributions but fail to do so, for groups of size "n." The suggestion was made earlier that unpredictability to the subject is the basic requirement of psychoacoustic noise. On that basis, the pseudo random noise produced by the additive algorithm may be considered acceptable.

C. Rayleigh Distribution

The Rayleigh distribution is required as an intermediate step between the uniform and Gaussian distributions. The Rayleigh distribution is obtainable by an analytical operation on the uniformly distributed numbers and is given by

$$R(I) = (2\mathrm{Ln}(1/X(I))^{\frac{1}{2}}$$

where R(I) is a member of the Rayleigh distribution and X(I), a member of the uniform distribution. Since X(I) assumes values from "0" up to but not including "1", R(I) varies, theoretically, from infinity down to but not including "0." In practice, the maximum value of R(I) is limited by the range of the computer or peripherals.

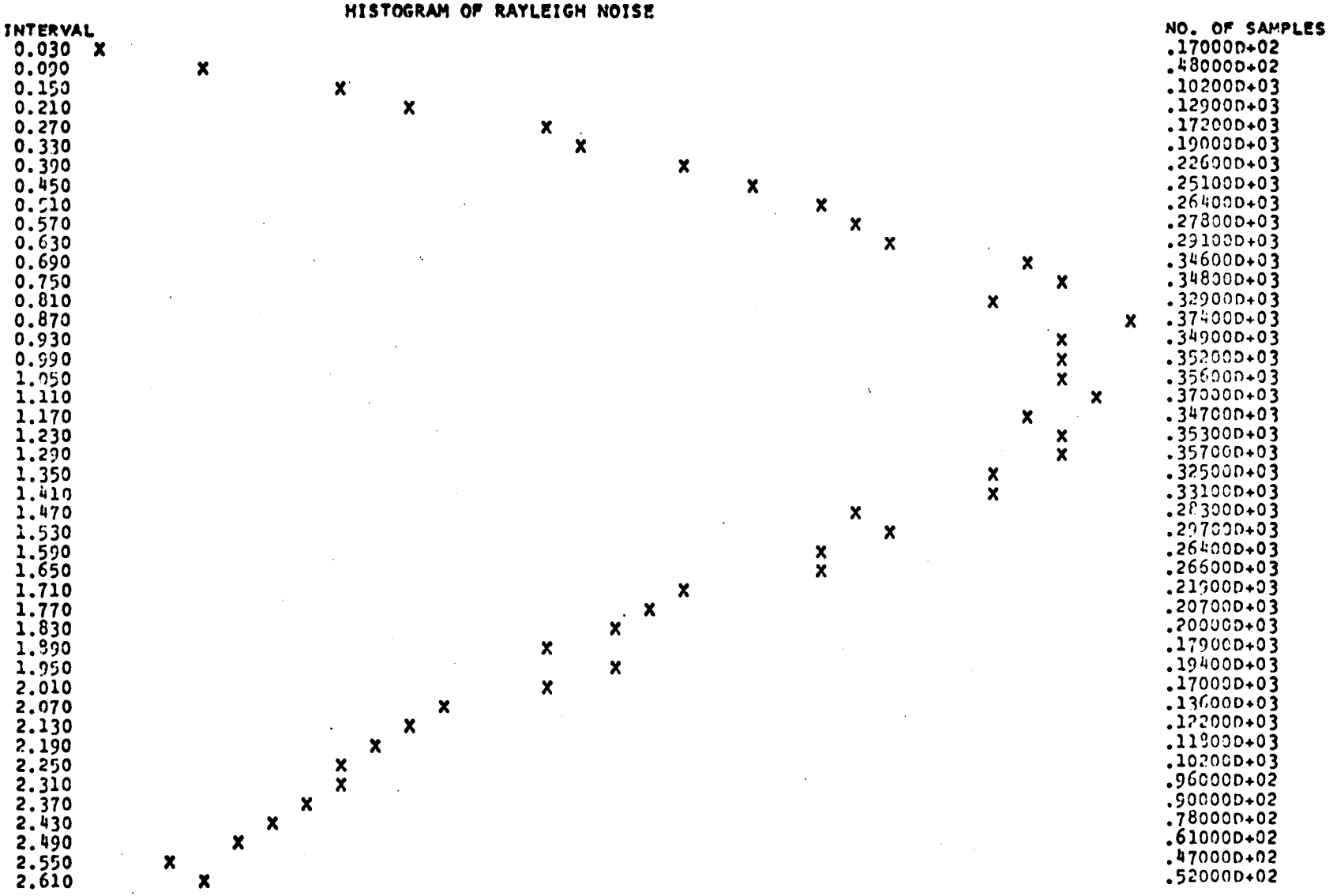
In nature, Rayleigh distributions arise, commonly, from envelope detection of narrow band Gaussian noise and, as a result, such Rayleigh signals change relatively slowly. No such restriction applies here. The signals may change at any rate subject only to the cycling rate of the computer. Figure 6 displays the histogram of the Rayleigh distribution.

D. Gaussian Distribution

The Rayleigh probability density distribution is

FIGURE 6 Histogram of Rayleigh Noise

The abscissa values given under "interval" bracket the interval within which the noise values are summed. The ordinate values given under "no. of samples" give the sum plotted across from the upper end of the interval.


HISTOGRAM OF RAYLEIGH NOISE
INTERVAL
NO. OF SAMPLES
0.030 .17000D+02
0.090 .48000D+02
0.150 .10200D+03
0.210 .12900D+03
0.270 .17200D+03
0.330 .19000D+03
0.390 .22600D+03
0.450 .25100D+03
0.510 .26400D+03
0.570 .27800D+03
0.630 .29100D+03
0.690 .34600D+03
0.750 .34800D+03
0.810 .32900D+03
0.870 .37400D+03
0.930 .34900D+03
0.990 .35200D+03
1.050 .35600D+03
1.110 .37000D+03
1.170 .34700D+03
1.230 .35300D+03
1.290 .35700D+03
1.350 .32500D+03
1.410 .33100D+03
1.470 .28300D+03
1.530 .29700D+03
1.590 .26400D+03
1.650 .26600D+03
1.710 .21900D+03
1.770 .20700D+03
1.830 .20000D+03
1.890 .17900D+03
1.950 .19400D+03
2.010 .17000D+03
2.070 .13600D+03
2.130 .12200D+03
2.190 .11900D+03
2.250 .10200D+03
2.310 .96000D+02
2.370 .90000D+02
2.430 .78000D+02
2.490 .61000D+02
2.550 .47000D+02
2.610 .52000D+02

converted to Gaussian by a direct transformation technique (Box and Muller, 1958 and Gold and Rader, 1969). The transformation technique relies upon the fact that the probability of the transformed number group equals the probability of the same group prior to transformation. The probability of occurrence of a number lying between an upper and lower bound or of two numbers lying within two sets of bounds, is given by the area under a two dimensional density distribution or the volume under a joint distribution, bounded by the bounds. This area, or volume survives transformation. For the latter case, a joint distribution, this relationship is given by

1. $\int\int p(x,y)dxdy = \int\int q(x',y')dx'dy'$

or when the limits of integration are equal as in the present case.

2. $p(x,y)dxdy = q(x',y')dx'dy'$

where $p(x,y)$ is the original probability density distribution and $q(x',y')$ is the transformed distribution. It will now be shown that the expressions

3. $x(I) = R(I)\sin(2\pi U(I-1))$

4. $x(I+1) = R(I)\cos(2\pi U(I-1))$

when interleaved (arranged in ascending $I = 1,3,5\ldots.n$) generate a Gaussian probability density distribution of

which x(I) and x(I+1) are successive members of the distribution and R(I) and U(I-1) are respectively members of independent Rayleigh and uniform density distributions, respectively.

The expressions on the right side of 3 and 4 are members of the joint distribution

5. $p(R,\theta) = p(R)p(\theta)$

since "R" and "θ" are members of independent distributions. The Rayleigh distribution, p(R), is given by

6. $p(R) = R \cdot \mathrm{Exp}(-R^2/(2\sigma^2))/\sigma$

where σ is a parameter called variance. The expression for the uniform density distribution, for this case is

7. $p(\theta) = p(2\pi U) = 1/2\pi$

indicating that "θ" varies uniformly from "0" up to but not including "2π" with an expected value of $1/2\pi$. Therefore

8. $p(R,\theta) = p(R)p(\theta) = R \cdot \mathrm{Exp}(-R^2/(2\sigma^2))/(2\pi\sigma)$

Equating this distribution to its target transformation yields

9. $p(R,\theta)dRd\theta = R \cdot \mathrm{Exp}(-R^2/(2\sigma^2))/(2\pi\sigma)dRd\theta$

$= p(x,y)dxdy$

where $P(R,\theta)$ in its explicit form, is the distribution arising from expressions 5 through 8, and where 9 and 10 result from substituting these equations in 2.

The transformations given in 3 and 4 may be written as

10. $X_1 = R \cdot \cos\theta$

11. $X_2 = R \cdot \sin\theta$

which gives

12. $R^2 = X_1^2 + X_2^2$

which are familiar ones to mathematicians being the transformation from rectangular to polar coordinates. For this transformation

13. $dxdy = RdRd\theta.$

Substituting this transformation in 9 gives

14. $p(x,y)RdRd\theta = Exp(-R^2/(2\sigma^2))/(2\pi\sigma)RdRd\theta$

and

15. $p(x,y)=Exp(-R^2/(2\sigma^2))/(2\pi\sigma)$

$=Exp(-(X^2+Y^2)/(2\sigma^2))/(2\pi\sigma)$

which is the Gaussian distribution in terms of X and Y or as required in 3 and 4, in terms of X(I) and X(I+1). It is seen in addition that

16. $p(x,y)=Exp(-(X^2+Y^2)/(2\sigma^2))/(2\pi\sigma)$

$=Exp(-X^2/2\sigma^2))/(2\pi\sigma)+Exp(-Y^2/(2\sigma^2))/(2\pi\sigma)$

$= p(X)p(Y)$

which shows that the transformation given by 3 and 4 produces two independent Gaussian distributions. When these two distributions are interleaved the individual

FIGURE 7: Histogram of Gaussian Noise

The abscissa values given under "interval" bracket the interval within which the noise values are summed. The ordinate values given under "no. of samples" give the sum plotted across from the upper end of the interval.

HISTOGRAM OF GAUSSIAN NOISE

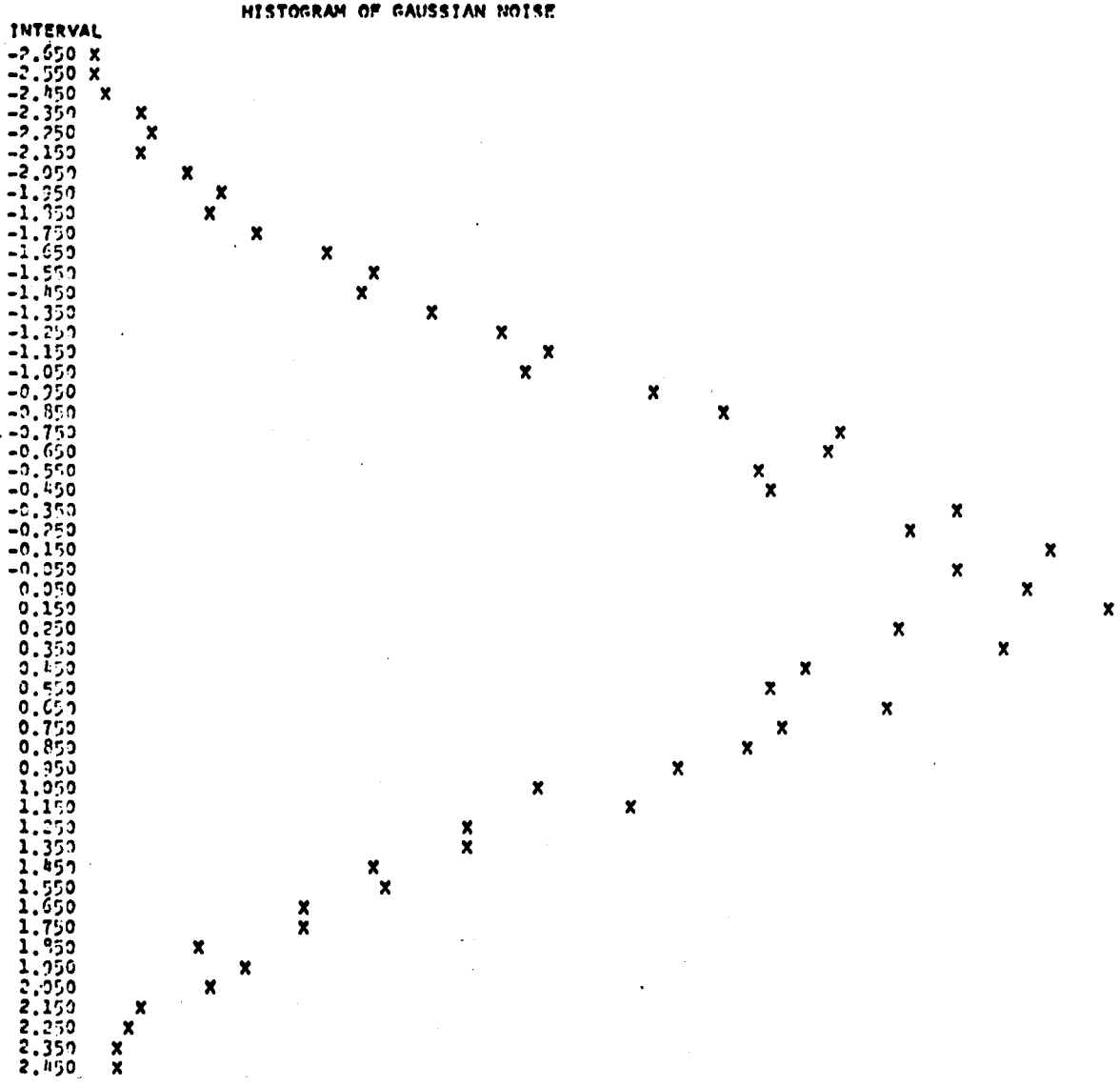
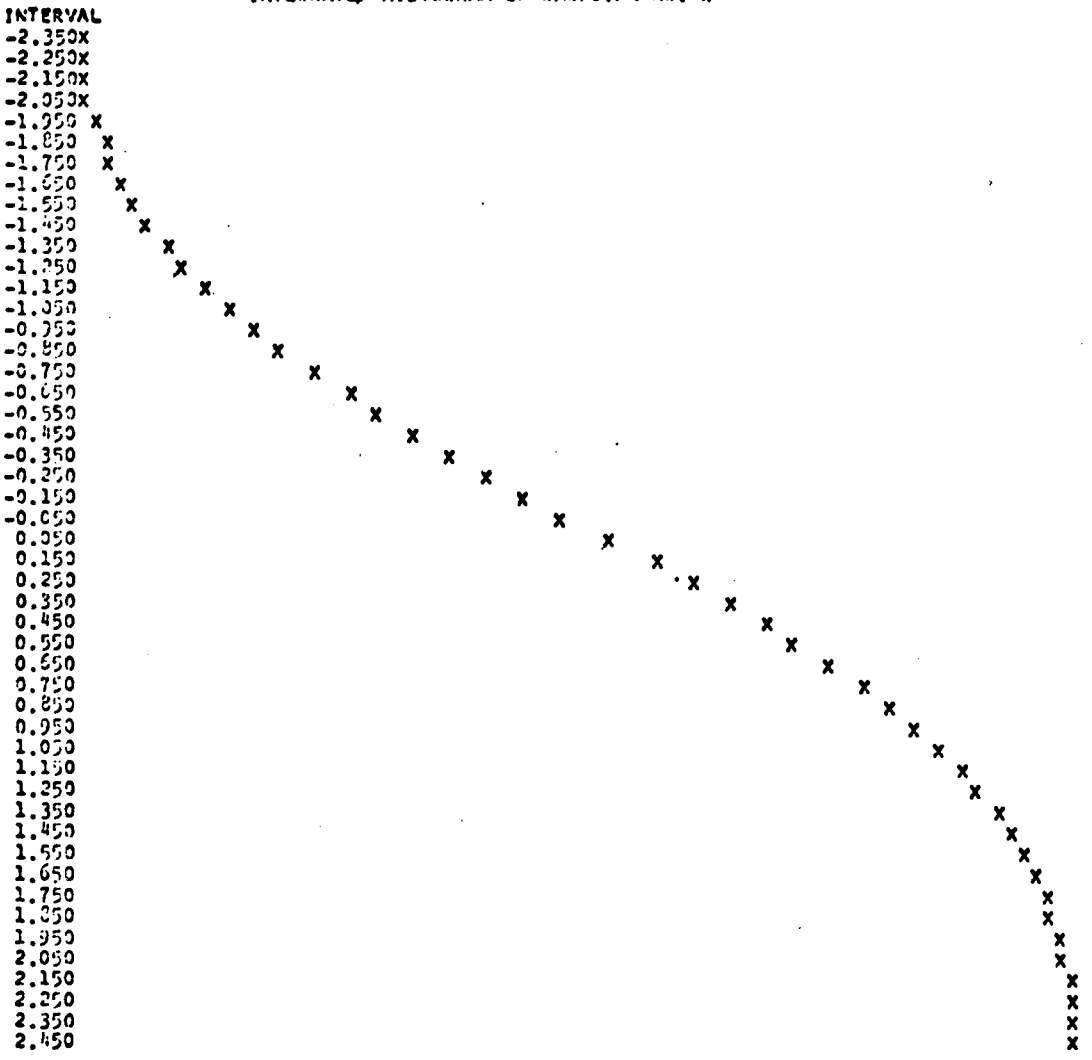


FIGURE 8: Integrated Histogram of Gaussian Noise

The abcissa values given under "interval" represent the upper value to which the noise is summed. The lower end of the interval is always minus infinity. The ordinate value given under "cumulative samples" is the integrated sum.

INTEGRATED HISTOGRAM OF GAUSSIAN NOISE



INTERVAL	CUMULATIVE SAMPLES
-2.350	.11400D+03
-2.250	.15200D+03
-2.150	.18400D+03
-2.050	.23600D+03
-1.950	.30200D+03
-1.850	.36800D+03
-1.750	.44600D+03
-1.650	.56000D+03
-1.550	.69400D+03
-1.450	.82000D+03
-1.350	.97600D+03
-1.250	.11080D+04
-1.150	.13720D+04
-1.050	.15700D+04
-0.950	.18020D+04
-0.850	.21080D+04
-0.750	.24400D+04
-0.650	.27600D+04
-0.550	.30660D+04
-0.450	.33660D+04
-0.350	.37460D+04
-0.250	.41020D+04
-0.150	.45300D+04
-0.050	.49140D+04
0.050	.53280D+04
0.150	.57740D+04
0.250	.61320D+04
0.350	.66320D+04
0.450	.69580D+04
0.550	.71520D+04
0.650	.75060D+04
0.750	.78120D+04
0.850	.81060D+04
0.950	.83600D+04
1.050	.85680D+04
1.150	.88100D+04
1.250	.89800D+04
1.350	.91500D+04
1.450	.92880D+04
1.550	.94160D+04
1.650	.95160D+04
1.750	.96280D+04
1.850	.96780D+04
1.950	.97540D+04
2.050	.98160D+04
2.150	.98460D+04
2.250	.98740D+04
2.350	.98960D+04
2.450	.99200D+04

occurrence of each number is doubled as is the total sample size. Therefore all probabilities remain unchanged and the resultant distribution is Gaussian with an expected value of "0" and a variance of "σ^2." In terms of electrical measures, "σ" is the RMS value of the noise.

Figures 7 and 8 show a histogram and an integrated histogram (the probability density distribution and the cumulative probability distribution) for 10,000 samples. The results clearly approximate the expected curves. A maximum likelihood fit Gaussian curve was fitted to Figure 8 by probit analysis and the goodness of fit measured in terms of chi squared. A chi squared of 0.028 at a df of 12 was calculated. The variance of the Gaussian curve was "1".

E. Long Duration Sinusoid

The Algorithm

Computers typically, provide internal functions for the generation of sinusoidally changing numbers derived from the series

$$\sin X = X - X^3/3! + X^5/5! - X^7/7! + \ldots (-1)^{n-1} X^{2n-1}/(2n-1)! + \ldots$$

where "n" is the number of the term and "X" is the angle in radians. This algorithm is poor for large "X" since increasing "X" requires an increased number of terms to maintain accuracy, thus leading to prohibitive computing times. For special cases where the frequency of the sinusoid

is an integer sub-multiple of the program's sampling frequency, one cycle of sinusoid may be generated, read into memory, and reread the desired number of times. This technique is used for the generation of a 20 second calibrating tone of 1000 Hz but it is unsuitable for the general case.

The algorithm which generates the sinusoidal tone for the signal-noise presentations, is based on the difference equations

$$Y(I) = Y(I-1)\cos(bT) + X(I-1)\sin(bT)$$

$$X(I) = -Y(I-1)\sin(bT)+X(I-1)\cos(bT)$$

where "Y" and "X" are each sinusoidal numbers; "I" is the current cycle, "b" is the radian frequency, and "T" is the cycling period. It will be instructive to consider a few cycles of the algorithm for the case where $Y(0) = 0$ and $X(0) = 1$ are established as initial conditions. Then

I=1 $\quad Y(1)=0\cdot\cos(bT)+\sin(bT) = \sin(bT)$

$\quad X(1)=-0\cdot\sin(bT)+\cos(bT)= \cos(bT)$

I=2 $\quad Y(2)=\sin(bt)\cdot\cos(bT)+\cos(bT)\cdot\sin(bT) = \sin(2bT)$

$\quad X(2)=-\sin(bT)\cdot\sin(bT)+\cos(bT)\cdot\cos(bT)= \cos(2bT)$

I=3 $\quad Y(3)=\sin(2bT)\cdot\cos(bT)+\cos(2bT)\cdot\sin(bT)=\sin(3bT)$

$\quad X(3)=-\sin(2bT)\cdot\cos(bT)+\cos(2bT)\cdot\sin(bT)=\cos(3bT)$

and in the general case

I=n $Y(n)=\sin(b(n-1)T)\cdot\cos(bT)+\cos(b(n-1)T)\cdot\cos(bt)$

$=\sin(nbT)$

$X(n)=-\sin(b(n-1)T)\cdot\sin(bT)+\cos(b(n-1)T)\cdot\cos(bT)$

$=\cos(nbT)$

This algorithm is described by Gold and Rader (1969).

Phase Considerations

Interest in the phase of a pure tone relative to the noise is concerned with starting phase, the relationship between the pure tone and that of the coincident noise. There may also be interest in producing two pure tones with fixed or varying phase, or finally, phase or frequency modulation may be required. (These capabilities are supported by the generation program.) For all cases the phase is most easily, and most smoothly, controlled by adding Y(n) to X(n) in appropriate ratios.

It is easily shown by trigonometric substitution that

$$\cos(bnT)+K\sin(bnT)=\sqrt{1+K^2}\cdot\cos(bnT-\theta)$$

where the phase angle "θ" is given by

$$\theta = \tan^{-1} K$$

$$K = \tan \theta$$

Noise Considerations

The unavoidable truncation of numbers by the computer, introduces an error which, on conversion to audio,

appears as noise. Gold and Rader (1969) show that, for the sinusoidal algorithm, the RMS value of this noise is given by

$$\sigma = \sqrt{nE_o^2/12}$$

where "n" is the number of iterations of the algorithm and "E_o" is the maximum error introduced by a single truncation. The magnitude of this error is taken as the numerical value of the lowest order bit. The magnitude of the noise voltage is, most usefully, calculated in bits relative to the maximum precision of the computer. Thus

$$(E_o')^2 = 2^{-2B}$$

where E_o' is the error relative to maximum precision and "B" is the maximum precision in binary bits. The ratio "n/12" is expressed as a power of 2 by

$$n/12 = 2^K$$

Then

$$\sigma' = \sqrt{2^{-2(B-K/2)}} = 2^{-(B-K/2)}$$

$$\log_2 \sigma' = -B+K/2$$

where "σ'" is the truncation noise due to the maximum sinusoid RMS and "$\log_2 \sigma'$" is this truncation noise to signal-ratio in bits. K/2. is an absolute measure of the signal degradation due to the noise.

For long signal durations on low precision computers,

truncation noise may become significant. For example, after 10^6 iterations K/2 is equal to approximately 9 bits. The noise to signal ratio for 30,20, and 10 bit machines is -21, -11, and -1 bit(s) respectively. This corresponds to RMS ratios of -134,-54, and 0 dB respectively.

For small, low precision machines, the increase of noise with iterations of the sinusoidal algorithm may be reduced by using the technique of resetting initial values described by Gold and Rader (1969). The resetting technique compensates only for in-step changes for X(n) and Y(n) but does not compensate for relative drifts in phase. It is based on the test for sine wave purity given by

$$X^2(n) + Y^2(n) = 1$$

for the pure sine-cosine case. If this is not true, due to noise buildup, then the condition may be imposed by multiplying X(n) and Y(n) by the reciprocal

$$1/\sqrt{X^2(n) + Y^2(n)}$$

whenever the noise to signal ratio has reached preset limits.

F. Filter Simulation

Design Approach

The design technique used for filter simulation, is based on the Butterworth stagger-tuned filter. These filters consist of two or more single-tuned filter stages arranged so that the output of one stage is the input to the next (cascaded). The design requires the determination of a center frequency and bandwidth for each stage so that, in combination, the desired over all pass band is achieved. The Butterworth design produces an optimally flat pass band with good phase response. The design technique is discussed fully by Uitjens (1953) or Angelo (1958).

Because of cascading, the approach accommodates itself to next state programming. Since the same subroutine, the single-tuned stage, is used and reused with changes in center frequency and bandwidth only, programming economy is achieved. Because increasingly selective, steepskirted pass bands result from increasing the number of single-tuned stages, flexibility of design is enhanced. Finally the use of cascaded single-tuned stages produced by second order algorithms rather than a single high order one, reduces the effects of truncation errors and generates less overall noise.

The Single-Tuned Stage

The algorithm for the single-tuned filter is based

on the impulse invariance design and is derived from the difference equation..

$$Y(nT) = Exp(-aT)\cdot\cos(bT)\cdot 2y(nT-T) - X(nT-T)$$
$$-Exp(-2aT)\cdot Y(nT-2T) + X(nT)$$

where "Y(nT-T)" is the output for cycle "n". "T" is the sampling period, "Y(nT-T)" and "Y(nT-2T)" are outputs produced one and two cycles prior to "Y(nT)", "X(nT)" and "X(nT-T)" are current and immediately preceding inputs to the filter, "b" is very closely related to the center frequency of the filter, and "a" is closely related to bandwidth. This equation is discussed by Gold and Rader (1969). An analogue filter has the characteristic of most readily passing electrical signals changing at some specific rate and attenuating signals with different rates of change. This algorithm acts, in a similar fashion, on a stream of input numbers, favoring those which exhibit a specific range of rates of change.

The algorithm has been tested in two ways, analytically and empirically.

First, the analytical expression for the magnitude of the output as a function of frequency for a sinusoidally changing input number stream, was used. This expression was given by Gold and Rader (1969) as

$$H(w) = \frac{(1+q^2-2q\cos(wT))^{\frac{1}{2}}}{((1+r^2-2r\cos((w-w_r)T)\ (1+r^2-2r\cos((w+w_r)T))^{\frac{1}{2}}}$$

where $q=r\cos(w_r T)$, $r=\mathrm{Exp}(-aT)$, $w_r=b$ center frequency of the filter, and w is the variable radian frequency. Figures 9 and 10 are plots of this function, for the single-tuned algorithm for two bandwidths, 0.001 and 320 Hz, both centered at 800 Hz. The algorithm was also tested for a variety of bandwidths and center frequencies.

A second empirical test was made in which a sinusoidal numeric stream at a frequency two bandwidths below center frequency is applied to the filter. A period long enough for all transients to have subsided, is permitted to pass and then the RMS of the next 100 cycles is found and plotted. The input frequency is incremented and the process repeated until the filter pass band has been scanned. This test is the computer equivalent of a swept frequency test on an analog filter and shows the simulation action of the algorithm. (Fig. 13 displays these results for a quadruple-tuned stage.)

The results of both tests illustrate the resonant action of the algorithm and its similarity to a single tuned analog.

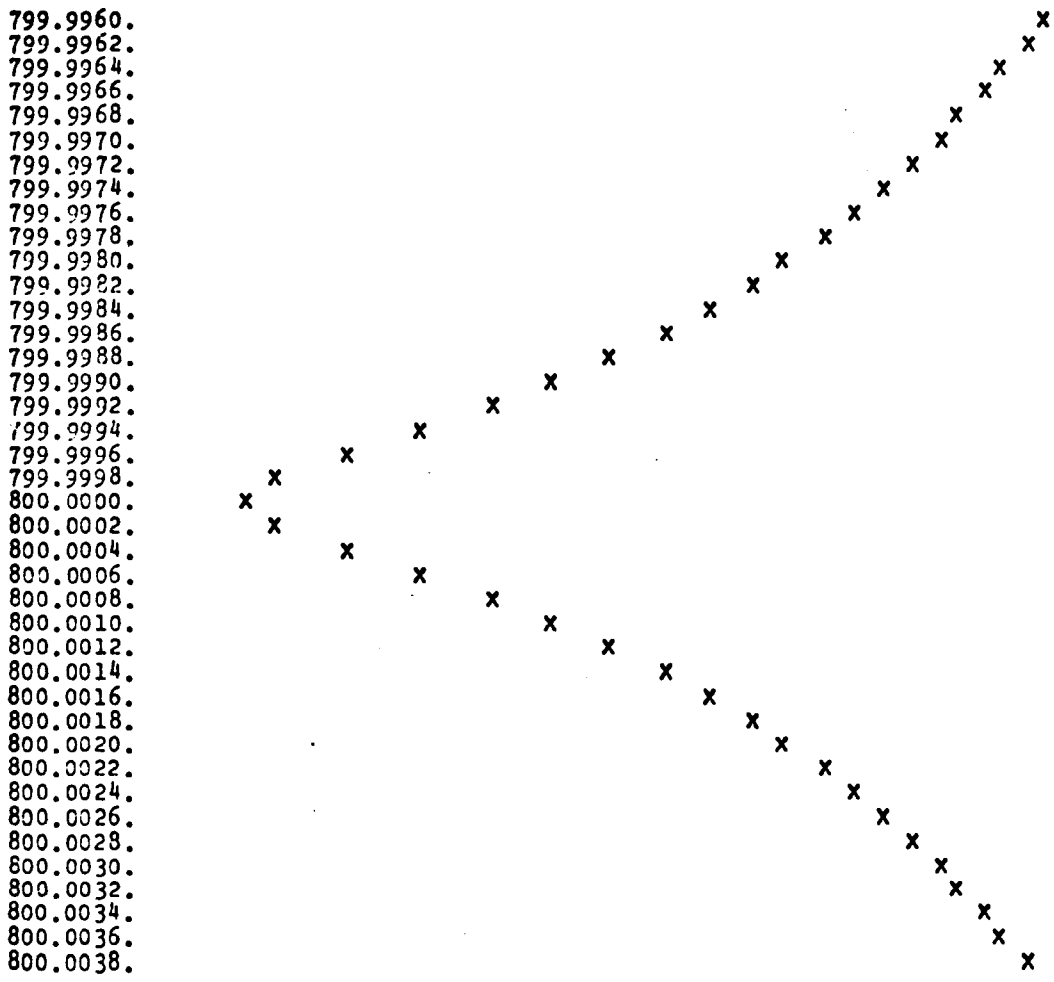
Butterworth Filters

The various Butterworth filters are simulated by

FIGURE 9: Frequency Response of Singly Tuned Stage: 0.001 Hz Bandwidth:

The abscissa values given under "frequency" are in hz. The ordinate values under "dB-out" give the attenuation of the filter in dB re: center frequency attenuation.

FREQUENCY RESPONSE OF SINGLY TUNED STAGE; CENTER FREQ.=800.000 BANDWIDTH= 0.001

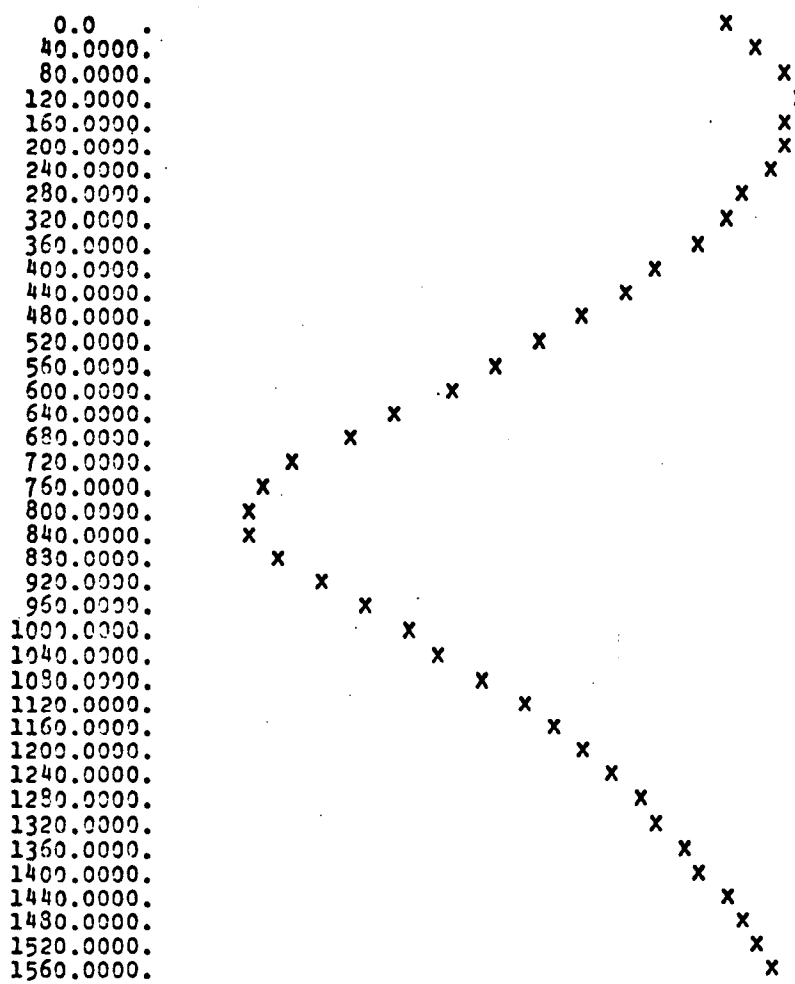


FREQUENCY	DB-OUT
799.9960.	18.12915
799.9962.	17.69078
799.9964.	17.22967
799.9966.	16.74318
799.9968.	16.22843
799.9970.	15.68203
799.9972.	15.10024
799.9974.	14.47769
799.9976.	13.80924
799.9978.	13.08802
799.9980.	12.30450
799.9982.	11.44906
799.9984.	10.50795
799.9986.	9.46466
799.9988.	8.30045
799.9990.	6.98970
799.9992.	5.51472
799.9994.	3.87758
799.9996.	2.15154
799.9998.	0.65199
800.0000.	-0.00000
800.0002.	0.65198
800.0004.	2.15153
800.0006.	3.87758
800.0008.	5.51471
800.0010.	6.98969
800.0012.	8.30044
800.0014.	9.46465
800.0016.	10.50793
800.0018.	11.44904
800.0020.	12.30448
800.0022.	13.08800
800.0024.	13.80922
800.0026.	14.47767
800.0028.	15.10021
800.0030.	15.68200
800.0032.	16.22840
800.0034.	16.74315
800.0036.	17.22963
800.0038.	17.69074

FIGURE 10: Frequency Response of Singly Tuned Stage: 320 Hz Bandwidth:

The abscissa values given under "frequency" are in hz. The ordinate values given under "dB-out" give the attenuation of the filter in dB re: center frequency attenuation.

FREQUENCY RESPONSE OF SINGLY TUNED STAGE; CENTER FREQ.=800.000 BANDWIDTH=320.000



FREQUENCY	DB-OUT
0.0	10.36167
40.0000.	11.29035
80.0000.	11.79972
120.0000.	12.00897
160.0000.	11.98823
200.0000.	11.78512
240.0000.	11.43426
280.0000.	10.96108
320.0000.	10.38391
360.0000.	9.71529
400.0000.	8.96306
440.0000.	8.13144
480.0000.	7.22203
520.0000.	6.23530
560.0000.	5.17290
600.0000.	4.04206
640.0000.	2.86392
680.0000.	1.68848
720.0000.	0.61564
760.0000.	-0.19345
800.0000.	-0.55896
840.0000.	-0.39644
880.0000.	0.21035
920.0000.	1.09234
960.0000.	2.05936
1000.0000.	3.04269
1040.0000.	3.99395
1080.0000.	4.86418
1120.0000.	5.67908
1160.0000.	6.43099
1200.0000.	7.12484
1240.0000.	7.76625
1280.0000.	8.36076
1320.0000.	8.91345
1360.0000.	9.42983
1400.0000.	9.91086
1440.0000.	10.36300
1480.0000.	10.78825
1520.0000.	11.18924
1560.0000.	11.56823

calling the single tuned subroutine the required number of times with the required design parameters. For example, a double tuned pair is simulated by calling the single tuned subroutine once with its center frequency and bandwidth set at

$$CF1 = CF - BW/(2\sqrt{2}) \qquad CF2 = CF + BW/(2\sqrt{2})$$

$$BW1 = CF1 \cdot BW/(CF\sqrt{2}) \qquad BW2 = BW1$$

where CF equals the overall center frequency, BW is the 3 dB overall bandwidth, CF1 is the center frequency of the first stage and BW1 is its bandwidth. The output of this stage becomes the input to the second stage in which CF_2 and BW_2 are the center frequency and bandwidth of the second stage. The output of stage two is the desired filtered noise.

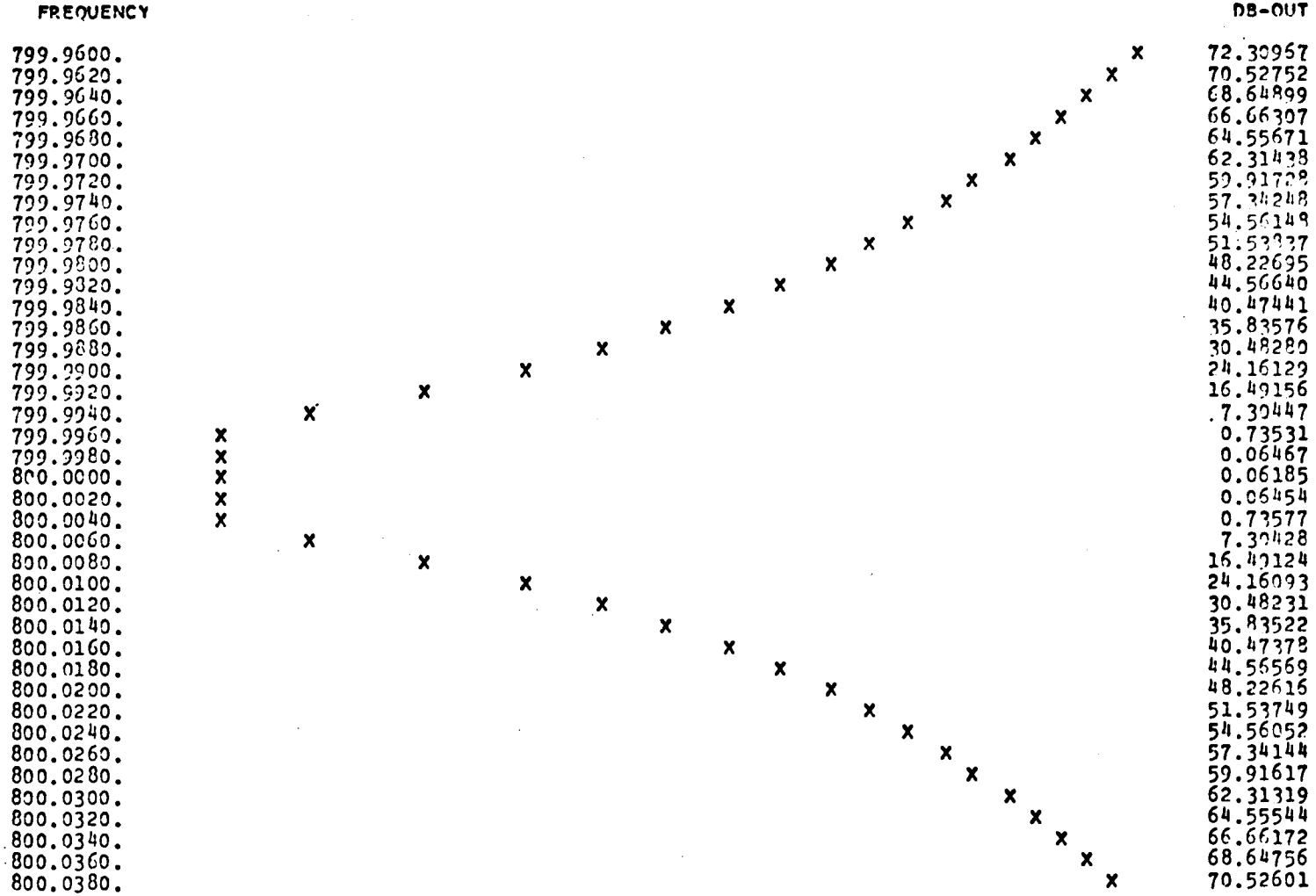
The same technique is used for triple, quadruple, sextuple, and octuple tuned filters, the required passes through the single tuned subroutine being three, four, six, and eight respectively. Figures 11 and 12 show the analytical responses for two quadruple tuned filters centered at 800 Hz with bandwidths of 0.01 and 320 Hz respectively. The test performed on the single tuned algorithms were also performed on the higher order filters for various bandwidths and center frequencies with satisfactory results.

As the passband is decreased or as the filter order

FIGURE 11: Frequency Response for Staggered Quadruple Stage: 0.01 Hz Bandwidth:

The abscissa values under "frequency" are in Hz. The ordinate values under "dB-out" give the attenuation of the filter in dB re: center frequency attenuation.

FREQUENCY RESPONSE FOR STAGGERED QUADRUPLE STAGE
CENTER FREQUENCY= 800.00 BANDWIDTH= 0.0100

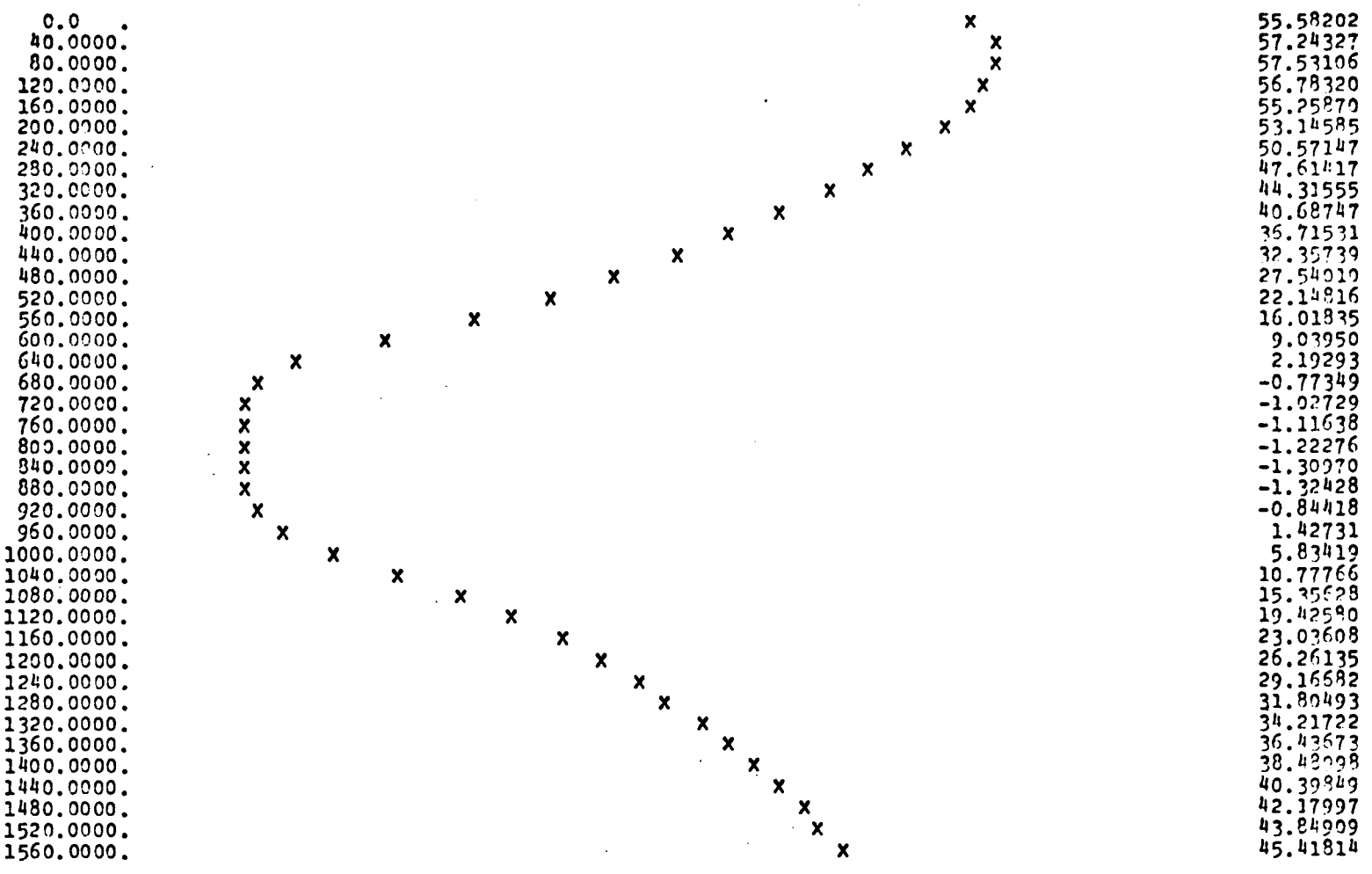


FREQUENCY	DB-OUT
799.9600.	72.30967
799.9620.	70.52752
799.9640.	68.64899
799.9660.	66.66307
799.9680.	64.55671
799.9700.	62.31438
799.9720.	59.91729
799.9740.	57.34248
799.9760.	54.56149
799.9780.	51.53337
799.9800.	48.22695
799.9820.	44.56640
799.9840.	40.47441
799.9860.	35.83576
799.9880.	30.48280
799.9900.	24.16129
799.9920.	16.49156
799.9940.	7.30447
799.9960.	0.73531
799.9980.	0.06467
800.0000.	0.06185
800.0020.	0.06454
800.0040.	0.73577
800.0060.	7.30428
800.0080.	16.49124
800.0100.	24.16093
800.0120.	30.48231
800.0140.	35.83522
800.0160.	40.47378
800.0180.	44.56569
800.0200.	48.22616
800.0220.	51.53749
800.0240.	54.56052
800.0260.	57.34144
800.0280.	59.91617
800.0300.	62.31319
800.0320.	64.55544
800.0340.	66.66172
800.0360.	68.64756
800.0380.	70.52601

FIGURE 12: Frequency Response for Staggered Quadruple Stage: 320 Hz Bandwidth:

The abscissa values given under "frequency" are in Hz. The ordinate values under "dB-out" give the attenuation of the filter in dB re: center frequency attenuation.

FREQUENCY RESPONSE FOR STAGGERED QUADRUPLE STAGE
CENTER FREQUENCY= 800.00 BANDWIDTH= 320.0000

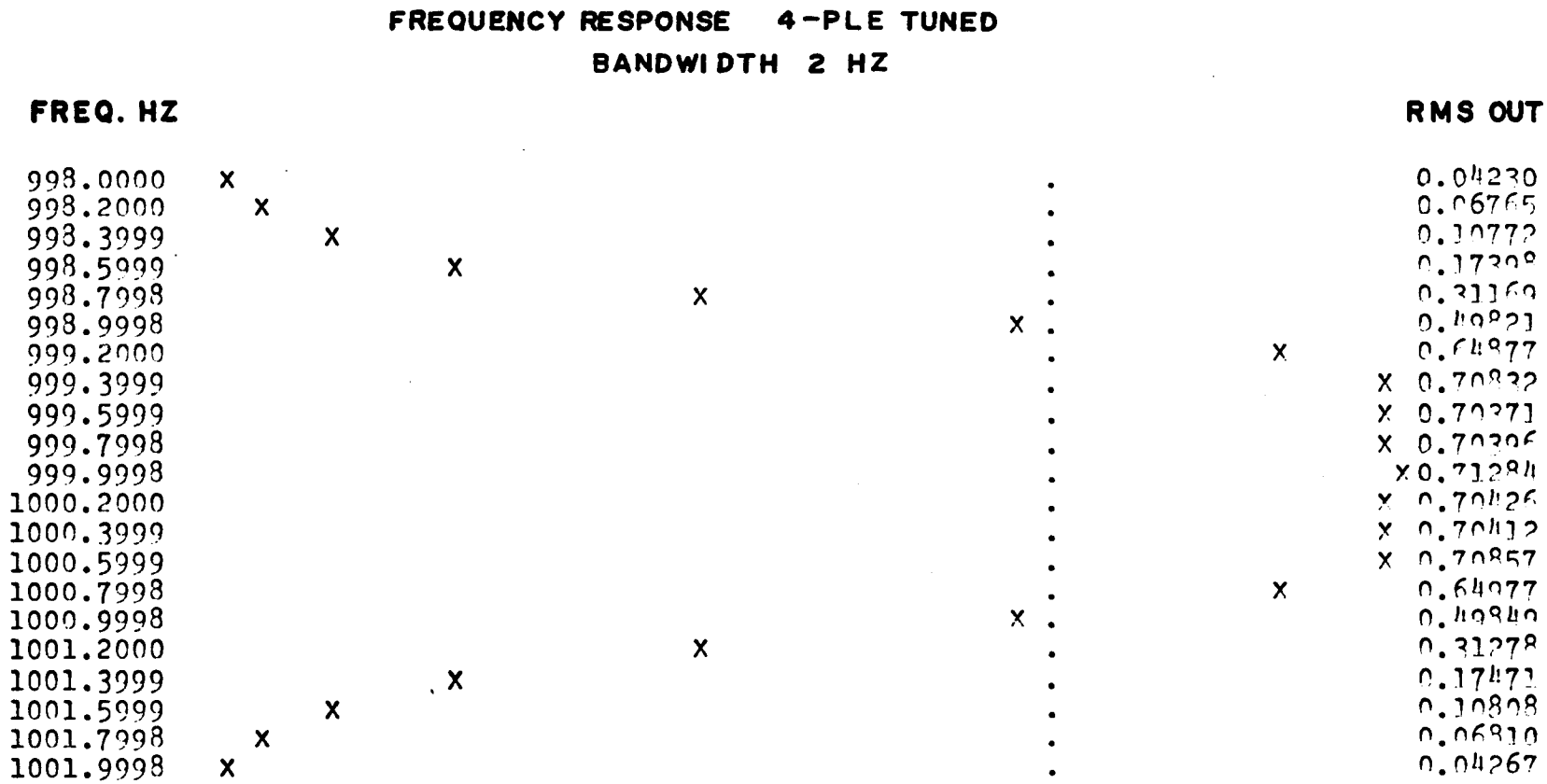


FREQUENCY	DB-OUT
0.0	55.58202
40.0000.	57.24327
80.0000.	57.53106
120.0000.	56.78320
160.0000.	55.25870
200.0000.	53.14585
240.0000.	50.57147
280.0000.	47.61417
320.0000.	44.31555
360.0000.	40.68747
400.0000.	36.71531
440.0000.	32.35739
480.0000.	27.54019
520.0000.	22.14816
560.0000.	16.01835
600.0000.	9.03950
640.0000.	2.19293
680.0000.	-0.77349
720.0000.	-1.02729
760.0000.	-1.11638
800.0000.	-1.22276
840.0000.	-1.30970
880.0000.	-1.32428
920.0000.	-0.84418
960.0000.	1.42731
1000.0000.	5.83419
1040.0000.	10.77766
1080.0000.	15.35628
1120.0000.	19.42590
1160.0000.	23.03608
1200.0000.	26.26135
1240.0000.	29.16682
1280.0000.	31.80493
1320.0000.	34.21722
1360.0000.	36.43673
1400.0000.	38.48098
1440.0000.	40.39849
1480.0000.	42.17997
1520.0000.	43.84909
1560.0000.	45.41814

FIGURE 13: Swept Frequency Response for the Staggered Quadruple Stage:

The abscissa values given under "Frequency" are in Hz. The ordinate values given under "RMS OUT" give the transmission characteristics of the filter.

FREQUENCY RESPONSE 4-PLE TUNED
BANDWIDTH 2 HZ



is increased, for a constant passband, the skirts fall off more and more steeply and the outermost tuned stages are required to be increasingly narrower. The lower limit of bandwidth, is a function of the precision of calculation. This lower limit has not been determined for double precision calculations (64 bits) but since a quadruple tuned filter with a bandwidth of 0.001 Hz has proven feasible, this lower limit may be ignored for most applications.

Quantization Effects

Since digital computers can only retain a limited number of significant figures in memory, truncation or rounding errors must appear. These errors significantly affect the generated signal and, therefore, require consideration. One type of error arises from the inexact values of the parameters of the algorithm and is anlogous to using inexact components in an analog filter. Gold and Rader (1969) show that this affects both center frequency and bandwidth. Moreover, as the sampling rate increases, the center frequency error also increases and is, in fact, directly proportional. The parameter error increases in significance as the center frequency is lowered and as the order of the difference equation increases, that is, as the parameter "n" in the expression

"(T-nT)" increases. The practical result of all this is to suggest the implementation of all filters as cascades or parallel connection of first or second order algorithms. This suggestion has been adhered to in this study.

The second type of error arises from the fact that the multiplication of two numbers of maximum precision produces a number of higher precision. It is therefore necessary to truncate each product. Since the error generated by each truncation is approximately random and assumed to be uniformly distributed, the effect is to introduce a noise with uniform probability density distribution into the final acoustic signal. This truncation effect may be viewed as individual sources of noise introduced at each product expression in the algorithm. Initially the noise is a function of the number of iterations but, typically approaches some fixed value. The overall noise RMS depends upon the number and position of these noise sources and therefore, depends upon the algorithm. The position within the algorithm in which the noise is introduced is important because it determines the subsequent processing. For example, noise introduced in the output circuit of an analogue amplifier is much less troublesome than noise at

the input since it is not amplified. Since filter algorithms amplify, this analogy is apt for integers or fixed point computer arithmetic but it is less apt for floating point arithmetic.

The digital resonator, used in the generation program has four sources of product noise. The analysis for the total RMS of the output noise is difficult but Gold and Rader (1969) provide a feasible analysis for three of the noises, these three being processed by the same transfer function. If the assumption is made that the fourth noise is of the same order, the analysis may be used for rough estimation. However, since Gold and Rader's work imply integer or fixed point arithmetic, some modification is required for the floating point case.

Examples of the three types of numbers are given below in decimal notation;

1.	Integers	134215	000013
2.	Fixed point	2.2142	0.0023
3.	Floating point	$.31478 \times 10^{10}$	$.31478 \times 10^{-5}$

Notice that for 1 and 2, precision is a function of magnitude but this is not so for 3, the floating point number. Therefore truncation to the lowest order digit produces a fixed maximum error "Eo for the integer and fixed point numbers but is a function of the power of 10, called

"characteristic", for this floating point case. In practice this means that while a fixed overall error noise may be calculated for cases "1" and "2", this is not possible for case "3". For the floating point case, it is more convenient to calculate an overall product noise to signal ratio (N/S) which varies as the signal mantissa (number without characteristic) goes from minimum to maximum but is unaffected by processes which affect only the mantissa. A knowledge of the specific architecture of the computer being used is required in these considerations.

The modified expression for the partial noise to signal ratio is given by

$$N/S = 2^{-(P-K/2)}$$

where "P" is the precision in bits and "K" is given by

$$K = \log_2 \frac{1}{12} \frac{1 + \mathrm{Exp}(-2\ BT)}{1-\mathrm{EXP}(-2\ BT)} \cdot \frac{1}{1-\mathrm{Exp}(-2\ BT)\cdot(4\mathrm{Cos}^2\ 2\ FT-2)+\mathrm{Exp}(-4\ BT)}$$

where "B" is the bandwidth in Hz, "T" is the sampling period, and "F" is the center frequency of the filter. The optimum results are shown in Tables 2 through 5 for various precisions. As the magnitude of the mantissa of the output deceases as a result of a change in input the "N/S" becomes worse to a degree dependent upon the

specific machine. These tables show the "N/S" for single tuned circuits only. When stages are cascaded to form a higher order filter the "N/S" will increase. A higher bound on this ratio may be formed by noting the N/S for the narrowest stage and adding 6dB per stage. This is a rough estimate of the partial noise only, but more precision is not justified. Tables 2 and 3 for 56 and 24 bit precision (the typical precision for 64 bit and 32 bit floating point machines) show insignificant noise to signal ratios for the worst case. Only very low bit machines give significant ratios, Tables 4 and 5. It should be noted, however, that attempts to produce narrow high order filters with many stages result in drastic increases in noise. It can also be shown that attempts to realize high order filters with a single algorithm produce significantly poorer "N/S", than by cascading.

The product noise estimation technique described above serves only to indicate a possible significance for the noise. If this possibility appears to exist then an empirical measure must be made. Familiarity with the arithmetic and number format for the specific computer, is needed to make this test. First a range of input signals is found which produce outputs with identical characteristics (exponents) but which vary over

TABLE 2

TRUNCATION NOISE FOR SINGLE-TUNED STAGE

PRECISION = 56 BITS

CENTER FREQ.	BANDWIDTH	NOISE/SIG.
800	.625000	-715.070
800	1.25000	-721.998
800	2.50000	-728.922
800	5	-735.837
800	10	-742.738
800	20	-749.608
800	40	-756.418
800	160	-769.621
800	320	-775.841
800	640	-781.860

TABLE 3

TRUNCATION NOISE FOR SINGLE-TUNED STAGE

PRECISION = 24 BITS

CENTER FREQ.	BANDWIDTH	NOISE/SIG.
800	.625000	-271.456
800	1.25000	-278.384
800	2.50000	-285.307
800	5	-292.223
800	10	-299.124
800	20	-305.993
800	40	-312.804
800	160	-326.006
800	320	-332.227
800	640	-338.246

TABLE 4

TRUNCATION NOISE FOR SINGLE-TUNED STAGE

PRECISION =	10 BITS	
CENTER FREQ.	BANDWIDTH	NOISE/SIG.
800	.625000	-77.3554
800	1.25000	-84.2829
800	2.50000	-91.2066
800	5	-98.1224
800	10	-105.023
800	20	-111.893
800	40	-118.703
800	160	-131.906
800	320	-138.126
800	640	-144.145

TABLE 5

TRUNCATION NOISE FOR SINGLE-TUNED STAGE

PRECISION = 8 BITS		
CENTER FREQ.	BANDWIDTH	NOISE/SIG.
800	.625000	-49.5708
800	1.25000	-56.4983
800	2.50000	-63.4220
800	5	-70.3378
800	10	-77.2381
800	20	-84.1080
800	40	-90.9185
800	160	-104.121
800	320	-110.341
800	640	-116.360

as wide a range of mantissas as possible. Assuming that, under these condition the output noise remains unchanged, then the output signals plus noise are given by

$$O_1^2 = (G \cdot I_1)^2 + N^2$$

$$O_2^2 = (G \cdot I_2)^2 + N^2$$

where "G" is the filter gain, "I_1" and "I_2" are the RMS input signals, and "N" is the RMS value of the product noise. Then

$$\frac{I_1^2}{I_2^2} = K^2$$

where "K" is the ratio of the extremes of the selected input range and finally

$$N^2 = \frac{(K \cdot O_2)^2 - O_1^2}{K^2 - 1}$$

Since the calculation of "K" is critical, a value of "I_2" should be chosen which is an integral multiple or sub-multiple of "I_1" for the computer used.

Input noise

The signal introduced into the filter is, in the case of the generation program, noise with a specified probability density distribution. This signal, however, is contaminated by the effects of quantization since an error is introduced in the lowest order digit by truncation.

This noise is uniformly assumed to be distributed and has an RMS given by

$$\sigma = Eo'/(12)^{\frac{1}{2}}$$

where Eo' is the error arising from truncation of the mantissa of the floating point number. When this noise, together with a signal, are processed by a filter algorithm, the signal to noise ratio at the output is not necessarily that which existed at the input. In the general case, the signal and noise at the output must be calculated separately. Gold and Rader (1969) give the equations for this analysis. However, in the case of the generation program both the desired and truncation noise have the same frequency spectrum, therefore, passage through the filter will not affect the signal to noise ratio. Calculations for 56, 24 and 10 bit precision give optimum signal-noise ratios of approximately -348, -145, and -72 dB respectively. If the signal at the output falls to the lowest end of the mantissa the signal-noise ratio will worsen to a degree dependent upon specific computer arithmetic but probably no more than 20 dB. Since the generation program was used in part 2 with 64 bit precision, internally 56 bits, this noise could be neglected.

G. Envelope of Pure Tone and Noise

Ensemble of Envelopes

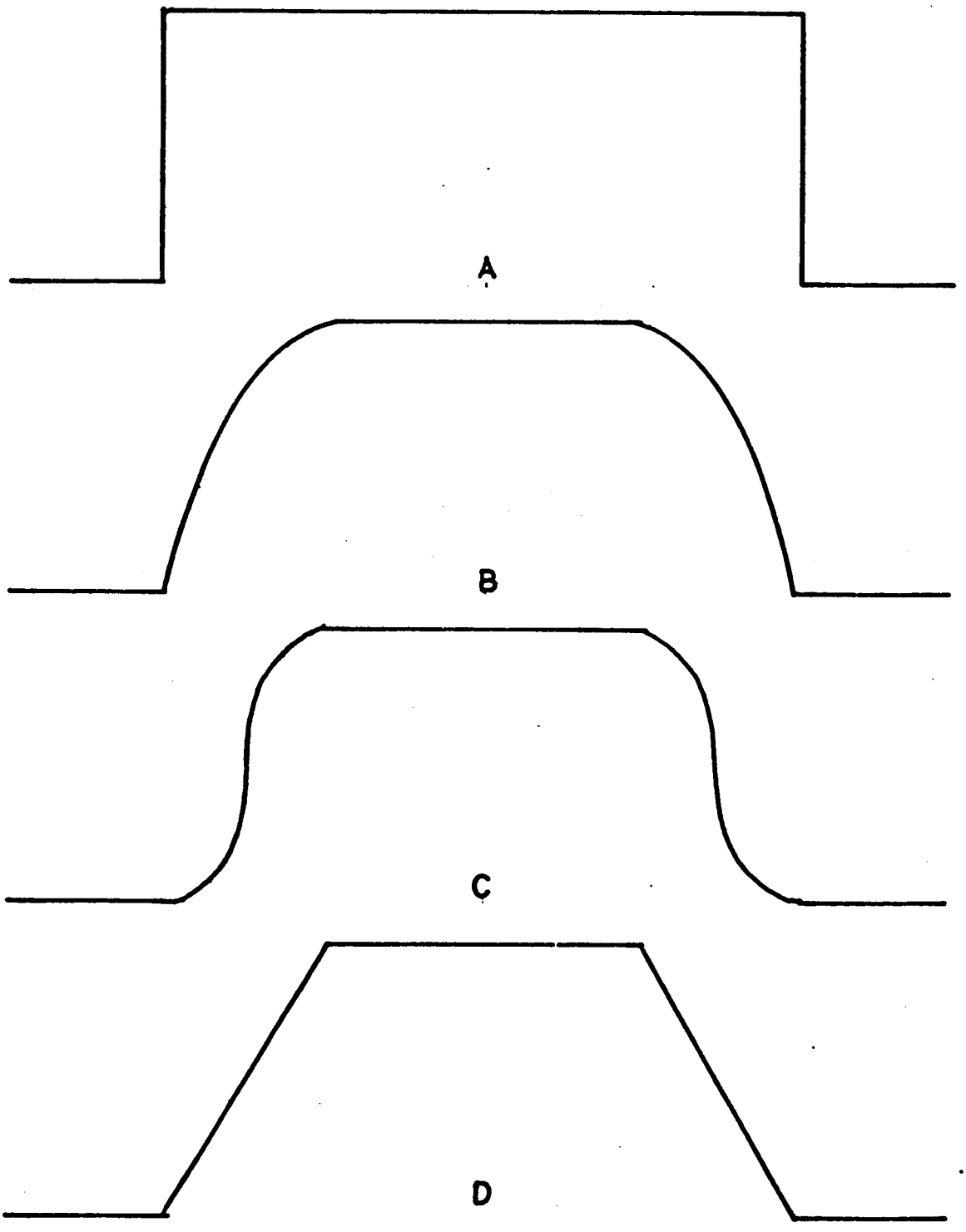
Four rise and fall algorithms are provided by the generation program for shaping the envelope of the tone and noise signals independently. They are step, sinusoidal, exponential, and ramp. With little programming effort, others can be added. These shaping functions may be combined as desired. For example, the noise signal may have a sinusoidal rise and a step fall while the pure tone has an exponential rise and a ramp fall. The envelope of a signal is determined by specifying rise shape and time, duration, and fall shape and time. The signal duration is measured from the beginning of the rise to the end of the fall. The rise and fall times of the ramp and sinusoidal shapes are measured from zero to maximum. For the exponential, the rise time is measured from zero to 95% of maximum, the fall time, from maximum to 5% of maximum. The four envelopes are shown in Figure 14 for the symmetrical case.

Spectral dispersion

In the generation program, the envelope of the noise is independent of the filter through which it passes. This is not so for hardware noise passed through an analogue filter. In this latter case, minimum rise and fall times are established by the filter characteristics. The program secures independence of the envelope by

FIGURE 14: Envelopes of Presentations.

Four symmetrical envelopes are shown. A, B, C, and D show "step", "exponential", "sinusoidal", and "ramp" rise and fall shapes, respectively.


A
B
C
D

allowing the filtered noise to reach the steady state before being shaped. This introduces flexibility but also requires the user to consider the spectral dispersion effects created by the choice of rise and fall times and shapes.

Figures 15 through 17 show these spectral components, for the step, sinusoidal, and ramp functions, respectively and for two rise time to duration ratios, .025 and 0.33 for symmetrical envelopes as a function of the normalized deviation from center frequency. The normalized deviation is given by

$$To\ (F-CF)$$

where "F" is the frequency in Hz., "CF" is the center frequency of the filter and "To" is closely related to duration and given by

$$To = D - Tr$$

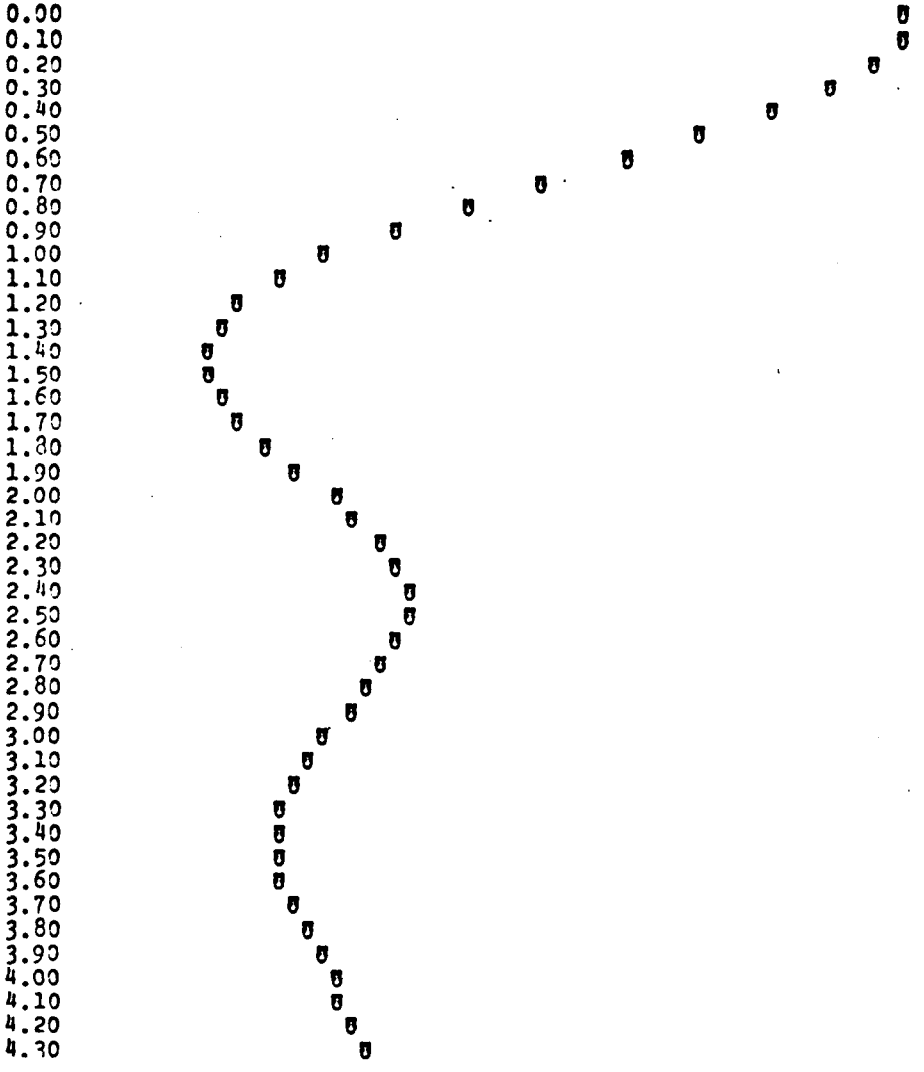
where "D" is the overall duration and "Tr" is the rise and fall time.

Figures 16 and 17 illustrate the decreases in spectral dispersion achieved by the ramp and sinusoidal functions for longer "Tr". Figures 18 through 20 show, quantitatively, the energy spill relative to total energy as a function of normalized deviation. For example, for a duration of .4 seconds, a center frequency of 800 hz

FIGURE 15: Spectral Dispersion for Step.

The abscissa given under "TO(F-CF)" is in normalized frequency units above center frequency. Both the "O" and the "*" plot the same curve for this special case. The ordinate, given under "O" and "*" give the ratio of the amplitude to maximum.

SPECTRAL DISPERSION FOR STEP

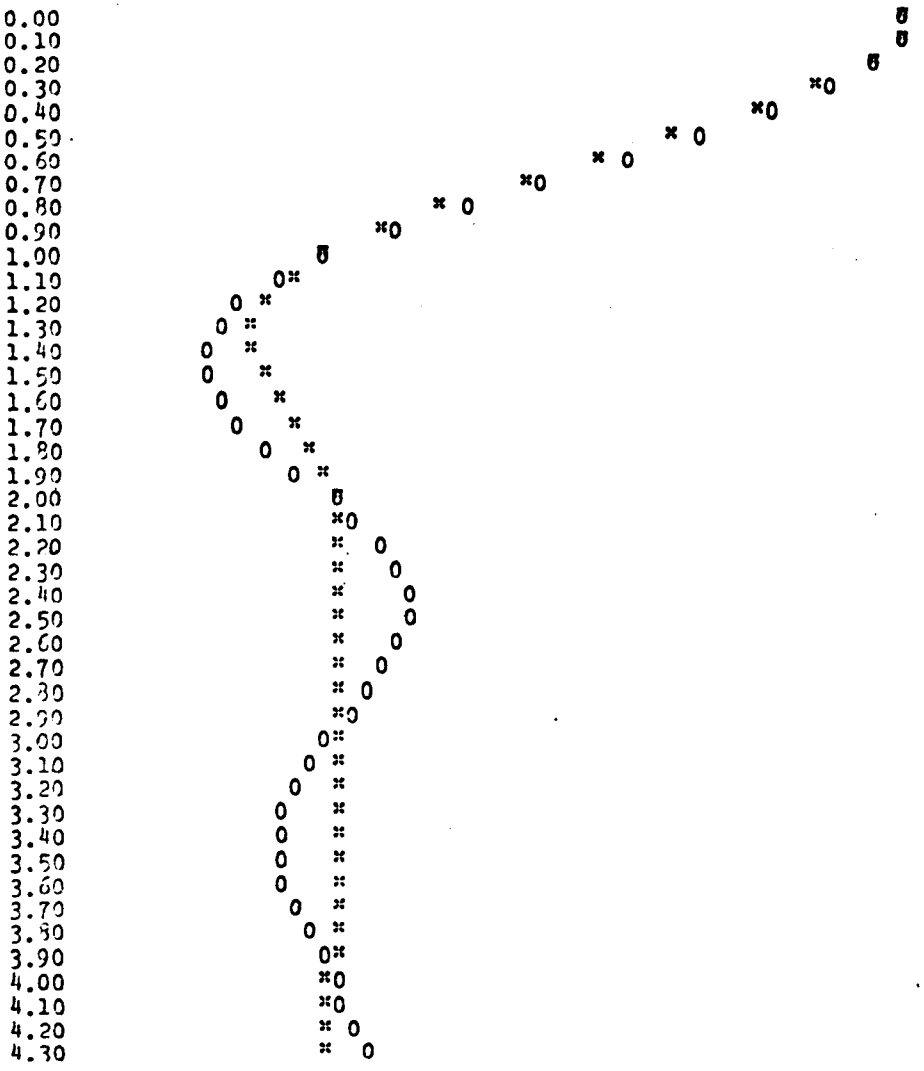


	RISE/DURATION
'0'=.000	'*'=.000
9.999974E-01	9.999974E-01
9.833041E-01	9.833041E-01
9.348554E-01	9.348554E-01
8.574903E-01	8.574903E-01
7.557064E-01	7.557064E-01
6.353461E-01	6.353461E-01
5.031951E-01	5.031951E-01
3.665180E-01	3.665180E-01
2.325692E-01	2.325692E-01
1.081149E-01	1.081149E-01
-9.984360E-04	-9.984360E-04
-9.020298E-02	-9.020298E-02
-1.564574E-01	-1.564574E-01
-1.983892E-01	-1.983892E-01
-2.163012E-01	-2.163012E-01
-2.120642E-01	-2.120642E-01
-1.888945E-01	-1.888945E-01
-1.510460E-01	-1.510460E-01
-1.034359E-01	-1.034359E-01
-5.124272E-02	-5.124272E-02
4.995190E-04	4.995190E-04
4.726968E-02	4.726968E-02
8.537274E-02	8.537274E-02
1.121705E-01	1.121705E-01
1.262132E-01	1.262132E-01
1.272724E-01	1.272724E-01
1.162707E-01	1.162707E-01
9.512376E-02	9.512376E-02
6.650763E-02	6.650763E-02
3.357887E-02	3.357887E-02
-3.331028E-04	-3.331028E-04
-3.202634E-02	-3.202634E-02
-5.870212E-02	-5.870212E-02
-7.818979E-02	-7.818979E-02
-8.910262E-02	-8.910262E-02
-9.091925E-02	-9.091925E-02
-8.398222E-02	-8.398222E-02
-6.942158E-02	-6.942158E-02
-4.901023E-02	-4.901023E-02
-2.497108E-02	-2.497108E-02
2.498736E-04	2.498736E-04
2.421697E-02	2.421697E-02
4.472877E-02	4.472877E-02
6.001035E-02	6.001035E-02

FIGURE 16: Spectral Dispersion for Sine.

The abscissa, given under "TO(F-CF)", is in normalized frequency units above center frequency. The ordinate, under "O" gives the ratio of the amplitude to maximum for rise and fall times equal to 2.5% of the duration. The ordinate under "*" is for rise and fall times equal to 33% of the duration.

SPECTRAL DISPERSION FOR SINE

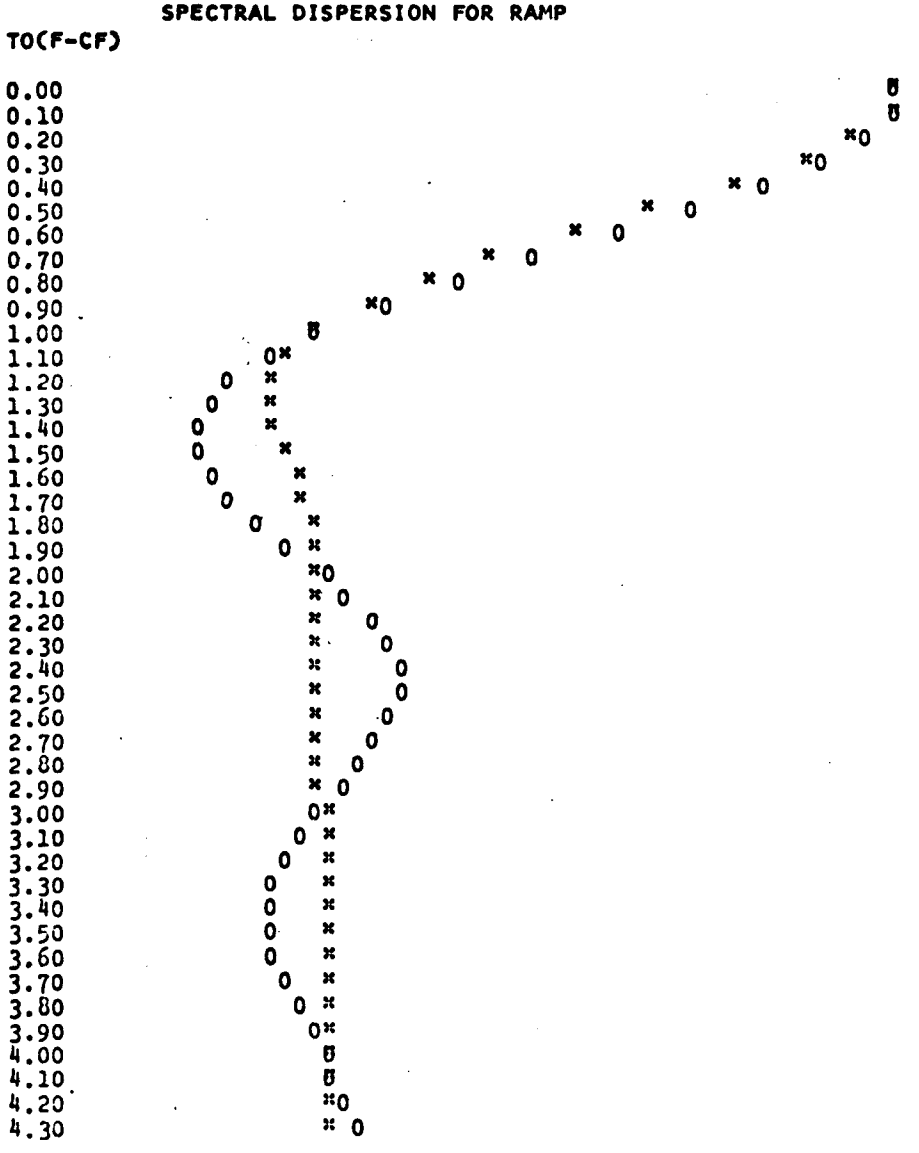


TO(F-CF)	'O'	RISE/DURATION '*'
0.00	9.999974E-01	9.999974E-01
0.10	9.832978E-01	9.809622E-01
0.20	9.348310E-01	9.260592E-01
0.30	8.574401E-01	8.394744E-01
0.40	7.556276E-01	7.276952E-01
0.50	6.352431E-01	5.988681E-01
0.60	5.030775E-01	4.620093E-01
0.70	3.664015E-01	3.261563E-01
0.80	2.324726E-01	1.995553E-01
0.90	1.080581E-01	8.897596E-02
1.00	-9.977887E-04	-7.835465E-04
1.10	-9.013223E-02	-6.715667E-02
1.20	-1.563114E-01	-1.098134E-01
1.30	-1.981719E-01	-1.304422E-01
1.40	-2.160265E-01	-1.323361E-01
1.50	-2.117550E-01	-1.198620E-01
1.60	-1.885813E-01	-9.787166E-02
1.70	-1.507633E-01	-7.113689E-02
1.80	-1.032189E-01	-4.386856E-02
1.90	-5.112295E-02	-1.936812E-02
2.00	4.982256E-04	1.662845E-04
2.10	4.713475E-02	1.367115E-02
2.20	8.510529E-02	2.110938E-02
2.30	1.117866E-01	2.325356E-02
2.40	1.257428E-01	2.140536E-02
2.50	1.267578E-01	1.709875E-02
2.60	1.157623E-01	1.182864E-02
2.70	9.467530E-02	6.833694E-03
2.80	6.617052E-02	2.987725E-03
2.90	3.339629E-02	7.013261E-04
3.00	-3.311650E-04	6.527278E-08
3.10	-3.182743E-02	5.872244E-04
3.20	-5.831369E-02	1.971305E-03
3.30	-7.763963E-02	3.597871E-03
3.40	-8.843725E-02	4.967086E-03
3.50	-9.019994E-02	5.720030E-03
3.60	-8.327943E-02	5.683906E-03
3.70	-6.880801E-02	4.875365E-03
3.80	-4.855342E-02	3.467921E-03
3.90	-2.472597E-02	1.735131E-03
4.00	2.472940E-04	-1.664935E-05
4.10	2.395436E-02	-1.511731E-03
4.20	4.421988E-02	-2.553873E-03
4.30	5.929486E-02	-3.053206E-03

FIGURE 17: Spectral Dispersion for Ramp.

The abscissa, given under "TO(F-CF)", is in normalized frequency units above center frequency. The ordinate, under "0" gives the ratio of the amplitude to maximum for rise and fall times equal to 2.5% of the duration. The ordinate under "*" is for rise and fall times equal to 33% of the duration.

SPECTRAL DISPERSION FOR RAMP

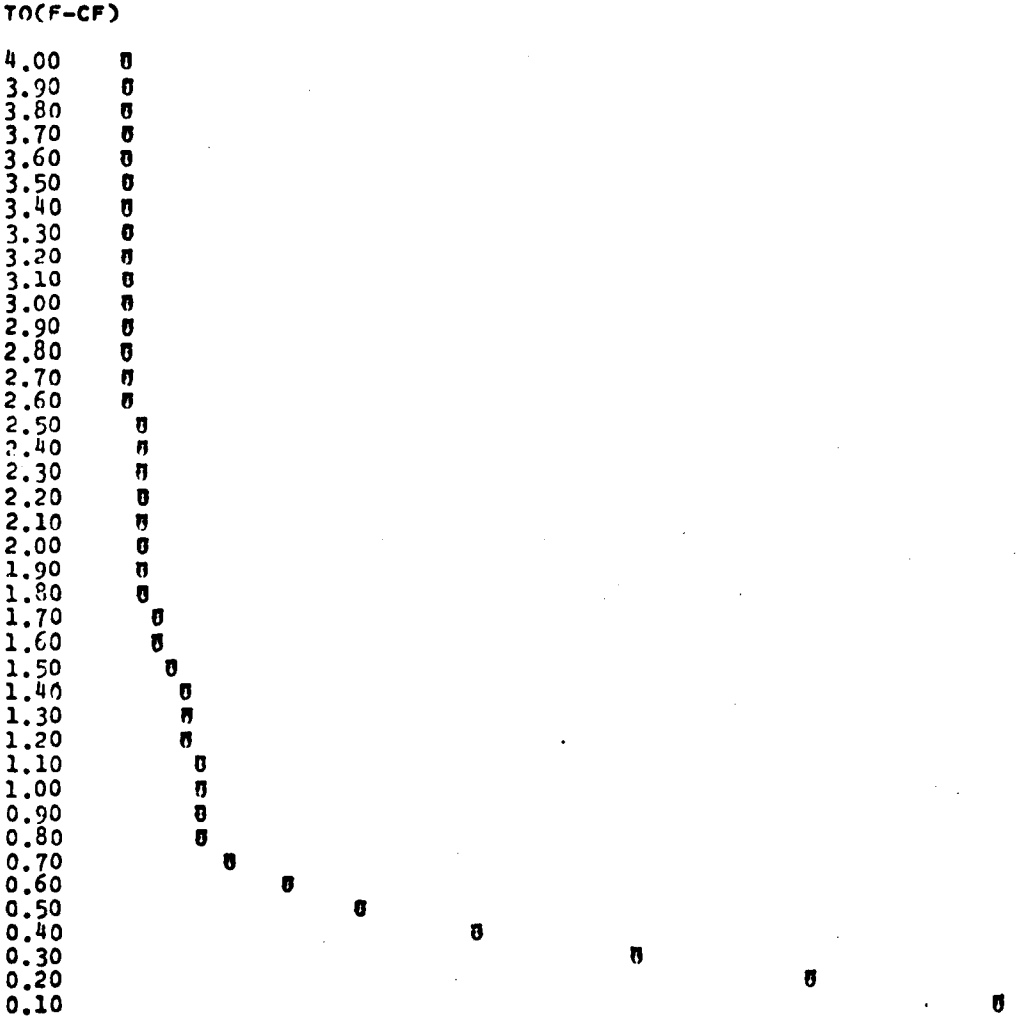


	RISE/DURATION	
	'O'	'*'
0.00	9.999969E-01	9.999966E-01
0.10	9.832919E-01	9.791836E-01
0.20	9.348116E-01	9.193999E-01
0.30	8.574013E-01	8.258969E-01
0.40	7.555674E-01	7.067161E-01
0.50	6.351645E-01	5.717670E-01
0.60	5.029876E-01	4.317125E-01
0.70	3.663124E-01	2.968146E-01
0.80	2.323992E-01	1.758044E-01
0.90	1.080149E-01	7.546901E-02
1.00	-9.972951E-04	-6.349080E-04
1.10	-9.007835E-02	-5.150214E-02
1.20	-1.562003E-01	-7.883477E-02
1.30	-1.980066E-01	-8.642792E-02
1.40	-2.158176E-01	-7.942610E-02
1.50	-2.115201E-01	-6.349939E-02
1.60	-1.883431E-01	-4.405417E-02
1.70	-1.505486E-01	-2.558541E-02
1.80	-1.030539E-01	-1.124391E-02
1.90	-5.103198E-02	-2.657915E-03
2.00	4.972433E-04	-2.493682E-07
2.10	4.703232E-02	-2.262820E-03
2.20	8.490234E-02	-7.667478E-03
2.30	1.114951E-01	-1.413267E-02
2.40	1.253859E-01	-1.971278E-02
2.50	1.263673E-01	-2.294353E-02
2.60	1.153765E-01	-2.304054E-02
2.70	9.433513E-02	-1.999274E-02
2.80	6.591475E-02	-1.438354E-02
2.90	3.325785E-02	-7.279906E-03
3.00	-3.296956E-04	7.066300E-05
3.10	-3.167664E-02	6.492279E-03
3.20	-5.801931E-02	1.109769E-02
3.30	-7.722276E-02	1.342510E-02
3.40	-8.793312E-02	1.347803E-02
3.50	-8.965498E-02	1.167204E-02
3.60	-8.274710E-02	8.708108E-03
3.70	-6.834334E-02	5.404591E-03
3.80	-4.820756E-02	2.524351E-03
3.90	-2.454043E-02	6.311812E-04
4.00	2.453420E-04	6.236120E-08
4.10	2.375564E-02	5.939118E-04
4.20	4.383492E-02	2.104682E-03
4.30	5.875374E-02	4.044994E-03

FIGURE 18: Energy Spill for Step.

The abscissa, given under "TO(F-CF)" is in normalized frequency units above center frequency. The ordinate, given under "O" and "*" plot the same curve and show the ratio of the energy relative to total energy which is spilled above the abscissa frequency.

ENERGY SPILL FOR STEP

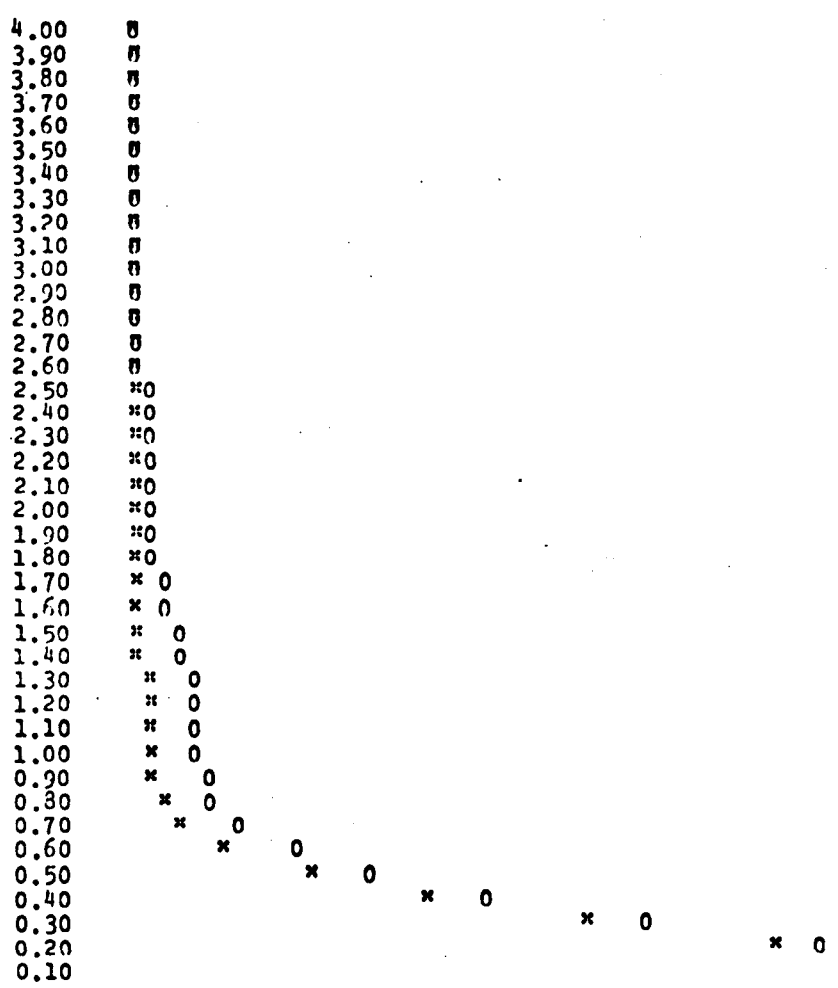


RELATIVE SPILL	
'0'=.000	'*'=.000
1.158614E-08	1.158614E-08
1.157222E-04	1.157222E-04
5.614522E-04	5.614522E-04
1.455759E-03	1.455759E-03
2.764559E-03	2.764559E-03
4.298504E-03	4.298504E-03
5.771763E-03	5.771763E-03
6.906248E-03	6.906248E-03
7.545694E-03	7.545694E-03
7.736027E-03	7.736027E-03
7.736049E-03	7.736049E-03
7.945284E-03	7.945284E-03
8.766088E-03	8.766088E-03
1.044518E-02	1.044518E-02
1.295381E-02	1.295381E-02
1.595964E-02	1.595964E-02
1.891564E-02	1.891564E-02
2.125045E-02	2.125045E-02
2.260294E-02	2.260294E-02
2.301758E-02	2.301758E-02
2.301762E-02	2.301762E-02
2.350489E-02	2.350489E-02
2.549025E-02	2.549025E-02
2.972392E-02	2.972392E-02
3.634509E-02	3.634509E-02
4.469021E-02	4.469021E-02
5.337215E-02	5.337215E-02
6.067570E-02	6.067570E-02
6.521815E-02	6.521815E-02
6.672709E-02	6.672709E-02
6.672817E-02	6.672817E-02
6.889724E-02	6.889724E-02
7.893419E-02	7.893419E-02
1.038622E-01	1.038622E-01
1.508485E-01	1.508485E-01
2.257531E-01	2.257531E-01
3.317274E-01	3.317274E-01
4.681721E-01	4.681721E-01
6.303483E-01	6.303483E-01
8.097697E-01	8.097697E-01

FIGURE 19: Energy Spill for Sine

The abscissa, given under "TO(F-CF)" is in normalized frequency units above center frequency. The ordinate, given under "O" and "*" show the ratio of energy relative to the total energy which is spilled above the abscissa frequency for rise and fall times relative to duration of 2.5% and 33%, respectively.

ENERGY SPILL FOR SINE

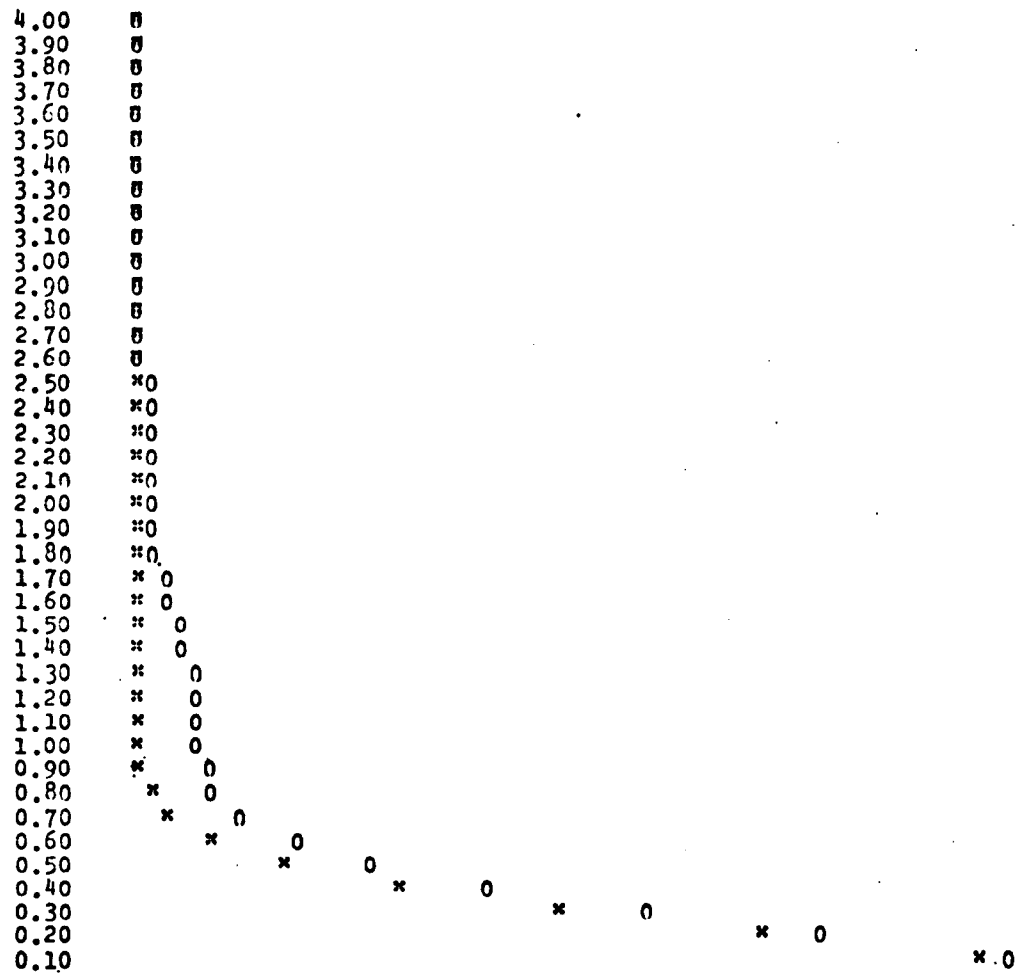


	RELATIVE SPILL	
	'0'	'*'
	1.135503E-08	5.697825E-11
	1.135302E-04	6.188979E-07
	5.512551E-04	3.090924E-06
	1.430357E-03	7.976644E-06
	2.719123E-03	1.461725E-05
	4.228807E-03	2.134253E-05
	5.681026E-03	2.641382E-05
	6.800279E-03	2.907458E-05
	7.431674E-03	2.987335E-05
	7.619764E-03	2.994423E-05
	7.619797E-03	2.994423E-05
	7.826875E-03	3.004553E-05
	8.639875E-03	3.188016E-05
	1.030419E-02	4.149320E-05
	1.279243E-02	7.025277E-05
	1.577582E-02	1.303483E-04
	1.871162E-02	2.245284E-04
	2.103187E-02	3.356742E-04
	2.237673E-02	4.272679E-04
	2.278923E-02	4.656852E-04
	2.278928E-02	4.656908E-04
	2.327457E-02	5.427971E-04
	2.525281E-02	9.383601E-04
	2.947320E-02	1.978527E-03
	3.607645E-02	3.947447E-03
	4.440229E-02	6.900548E-03
	5.306743E-02	1.050029E-02
	6.035942E-02	1.399760E-02
	6.488610E-02	1.647621E-02
	6.640452E-02	1.740333E-02
	6.640475E-02	1.740345E-02
	6.857281E-02	1.903071E-02
	7.860749E-02	2.721613E-02
	1.035349E-01	4.908194E-02
	1.505276E-01	9.205678E-02
	2.254532E-01	1.666753E-01
	3.314700E-01	2.755186E-01
	4.679812E-01	4.203721E-01
	6.302471E-01	5.966475E-01
*0	8.097748E-01	7.944440E-01

FIGURE 20: Energy Spill for Ramp

The abscissa, given under "TO(F-CF)" is in normalized frequency units above center frequency. The ordinate, given under "O" and "*" show the ratio of energy relative to the total energy which is spilled above the abscissa frequency for rise and fall times relative to duration of 2.5% and 33%, respectively.

ENERGY SPILL FOR RAMP



	RELATIVE SPILL	
	'0'	'*'
4.00	1.118161E-08	8.351700E-16
3.90	1.118841E-04	8.555673E-08
3.80	5.435931E-04	1.454059E-06
3.70	1.411259E-03	7.727008E-06
3.60	2.683198E-03	2.401223E-05
3.50	4.176367E-03	5.326989E-05
3.40	5.612734E-03	9.228195E-05
3.30	6.720509E-03	1.309882E-04
3.20	7.345836E-03	1.574374E-04
3.10	7.532231E-03	1.664893E-04
3.00	7.532253E-03	1.664903E-04
2.90	7.737722E-03	1.778718E-04
2.80	8.544821E-03	2.223019E-04
2.70	1.019795E-02	3.081420E-04
2.60	1.267077E-02	4.222381E-04
2.50	1.563716E-02	5.352902E-04
2.40	1.855766E-02	6.187432E-04
2.30	2.086689E-02	6.616369E-04
2.20	2.220595E-02	6.742624E-04
2.10	2.261687E-02	6.753620E-04
2.00	2.261691E-02	6.753620E-04
1.90	2.310069E-02	6.766794E-04
1.80	2.507352E-02	7.040300E-04
1.70	2.928393E-02	8.446108E-04
1.60	3.587343E-02	1.261399E-03
1.50	4.418465E-02	2.127331E-03
1.40	5.283701E-02	3.482123E-03
1.30	6.012017E-02	5.086306E-03
1.20	6.465250E-02	6.420999E-03
1.10	6.615984E-02	6.990633E-03
1.00	6.616002E-02	6.990723E-03
0.90	6.832736E-02	8.213581E-03
0.80	7.836031E-02	1.485720E-02
0.70	1.032869E-01	3.377707E-02
0.60	1.502845E-01	7.380241E-02
0.50	2.252259E-01	1.440100E-01
0.40	3.312742E-01	2.512666E-01
0.30	4.678360E-01	3.977531E-01
0.20	6.301698E-01	5.792858E-01
0.10	8.097779E-01	7.851941E-01

and a ramp rise and fall time of .1 seconds, the energy spill is .001% beyond 10 hz above or below center frequency. If the rise time is reduced to .01 seconds, the spill is .07%. The figures give only limiting conditions and it is not correct to interpolate linearly between the curves. If a specific investigation requires knowledge of the precise spill, commercial software for spectral analysis is available and, if accessible, should be used. The program user should note, however, that a quantitative knowledge of the energy spill does not establish the psychoacoustic effect. Defensible assumptions base on a knowledge of spill may be made, provided they are not closely related to the objectives of the investigation, but only experiment can securely establish these effects.

Algorithms

The rise and fall algorithms used in the generation program are given below:

1. Step - multiply by zero prior to turn on and after noise ends.
2. Exponential
   (a) Rise

   EXPR = A(1-Exp(-3(B-C)/D))

   (b) Fall

   EXPF = A Exp(-3(B-C)/D)

3. Sinusoidal

(a) Rise

RSIN = A•0.5(1-COS(.857 · 0.314593 · (B-C)/D))

(b) Fall

FSIN = .5•A(1 + COS(.857 · 0314593 · (B-C)/D))

4. Ramp

(a) Rise

RRAMP = A(B-C)/D

(b) Fall

FRAMP = A(1 - (B-(C-D))/D)

where "A" is the amplitude of the signal, "B" is the computation cycle, "C" is the starting cycle, and "D" is the rise or fall time as defined previously.

V. Output Considerations

General

The final products of the noise generation program prior to conversion to audio, are sets of numbers residing in peripheral memory, representing the desired output signal. If the cautions discussed in the previous section have been observed, the sets of numbers are relatively free of unwanted numerical error noises. The next step, typically, is to convert these numbers into audio signals and either record them on audio tape or deliver them directly to the subject's ear. This step

is particularly susceptible to the introduction of extraneous noise, partially because hardware is involved and partially because the high order of precision associated with computer calculation is lacking. These matters require consideration.

Digital to Analogue Conversation

The conversion of streams of numbers, represented by one of several types of binary codes, into analogue signals suitable for audition, is handled by a digital to analogue converter (DAC). Typically a DAC consists of a precise voltage source, a precise set of voltage dividers and a decoding and switching system that selects appropriate voltage dividers to produce a voltage proportional to the input binary stream. For example, a string of coded signals representing the numbers 2, 8, 10 may produce voltages of .02, .08, .10 volts. Though the DAC output is smoothed by filtering, it is, nevertheless, an incremental machine. An output number falling between two incremental output levels will, in some machines, produce the lower level. It therefore introduces quantization errors and audio noise. It was shown, previously, that the truncation error is dependent upon the bit precision of the system. Since DAC's are relatively low bit machines when compared to computers,

truncation noise can become significant. The truncation noise introduced by the DAC may be considered to be uniformly distributed and to have a variance given by

$$\sigma^2 = Eo^2/12$$

where "Eo" is the maximum error arising from truncation. For DAC's with overall precisions of 15, 12, and 8 bits, 14, 11, and 7 bits respectively, establish the value while the remaining bit establishes the sign. Truncation noise for such sign-magnitude converters is -95, -77, and -53 respectively, dB below the absolute value of the maximum integer handled by the system.

The optimum noise to signal ratio which can be realized depends upon the ratio of the RMS value of the signal to its peak. For example, the RMS of a sinusoidal wave is 3 dB below its peak so that the optimum noise to signal ratio is 3dB less than given above. For Gaussian noise in which the positive and negative excursions are limited to $\pm 4\sigma$, the optimum noise to signal ratio is 6dB below maximum values. When a pure tone is added to the Gaussian noise, provision must be made for the worst case. In Section 2 the pure tone plus Gaussian noise may reach a peak level approximately 20 dB above the RMS value of the noise alone. This gives an optimum noise to signal ratio of -75, -57, and -33 for the 15, 12, and 8 bit

machines. If the level of the noise is permitted to drop below its permissible maximum or if unnecessarily high signal to noise ratios are used then the conversion truncation noise may seriously degrade the output.

Though the sign magnitude conversion system which was the basis of the above calculations, is typical of machines presently in use, it must be understood that other systems are also in use. It is necessary that the user understand his particular machine particularly in regard to its contribution to total unwanted noise.

Taped Output

If the output of the DAC is stored on audio tape, consideration must be given to the effects of recording and playback. In addition to normal analogue amplifier noise, tape recorders are beset by problems of tape bounce, stretch, tape coating variations and imperfections, bias current distortion, print through, and variations in the speed of tape travel. In amplitude modulation (AM) recorders, bounce, coating variations, and bias current inbalance appear as a wide band noise produced by small changes in amplitude superimposed upon the signal. This noise, plus the amplifier noise, has two components, one an additive one which is independent

of the signal amplitude, the other, arising from a modulation of the signal itself. The first component is reducible by various filtering or cancellation strategems, the second is not. High grade to professional tape recorders have an overall wide band noise level 40 to 60 dB below the maximum recording level, though this figure is strongly dependent upon bias adjustment, the cleanliness and demagnetized condition of the heads, and the quality of the tape used. Frequency modulation (FM) tape recorders, typically used for instrumentation purposes, do not respond as readily to amplitude changes in recording and play back levels and may have a 10 dB lower noise level than the same recorder in an AM mode.

Variations in the speed of tape travel, produce effects in the 10 to 200 hz band termed wow and flutter and, sometimes, jitter. High grade tape recorders have wow and flutter deviations from average speed of less than .2% RMS. Instrumentation recorders in the FM mode have speed deviation of .03% peak to peak. The specifications quoted by manufacturers are average values only. It is not possible to predict the extent nor the nature of tape travel variations for short periods, but they are noticeable.

The investigator has the problem of determining the significance to his investigation of tape recorder disturbances. This must be done experimentally. Wow and flutter strongly affect center frequency but have much less effect upon the signal envelope. The psychoacoustic effect may depend strongly upon the duration of each presentation. A separate investigation in which directly generated signals are compared to recorded signals for what is judged to be, the most sensitive case, is in order.

VI. Evaluation

A. System Application

A system for generating pure tones and noise by computer has been described, above, together with its motivations and anticipated advantages. This system has been applied to a psychoacoustic investigation which will be discussed in Section 2 of this dissertation. This concluding section of Section 1 discusses and evaluates the functioning of this system based on this experience.

B. Preparation

The investigation in part 2 required 3200 trials, each consisting of two presentations, noise alone and noise plus pure tone. The complete investigation comprised 20 runs of 160 trials each, encompassed 10 bandwidths, and covered a range of 14 signal to noise ratios.

Within a run, the noise signal repeated itself, but for each bandwidth there were two runs, each one using a different noise sample. The programs were run on a Xerox Sigma 7 using punched card access.

For this investigation, the generation program operated in the double pass mode, the noise ensemble being generated in an initial pass, stored on digital tape and read back as needed for the second pass. For this first run, the investigator prepares data specifying the mode of operation, and the center frequency, filter type, duration, rise and fall shapes and times, and initial delay for each noise sample. In addition the signal to noise ratio is set at -200 dB and, though it has no effect on output, he specifies the pure tone envelope and frequency. Once the noise ensemble had been established, the necessary data preparation required minimal work for the investigator.

A second program was required to randomize both the signal to noise ratio and the interval in which the pure tone appeared within a run(for each run filter width and noise type were held constant). This program set up the data file used by the generation program during its second pass. The investigator must specify the number of trials and the ensemble of bandwidths, input noises, and

signal to noise ratios. For this investigation, the various ensembles having been established by a pilot run, the preparation of the data for the computer required only minutes.

When these initial preparations had been completed, the main program could be run. In the second pass the main program read data specification from the randomization program and selected the appropriate noise sample from the tape prepared by pass 1. The investigator only changes the mode and sets the starting and finishing numbers for the trial, a few minutes work.

The initial preparation for the investigation of Section 2 went as smoothly as implied above. With the assistance of a part time programmer and normal computer facility service, it should progress smoothly for even non-computer oriented investigators.

B. Production of Audio Tapes

The output of the digital to analogue converter was stored on audio tape for the investigation in Section 2. The investigator has no function here, except to check the final product. There were, however, a number of important practical considerations which emerged.

Each run of this investigation, had a real time duration of 20 minutes. The computer time totalled

approximately 23 minutes of which 6 minutes was spent accessing the input and output tapes. In a previous pilot run in which a different randomizing scheme required a search of the noise sample tape after each trial, the access time was between 40 to 50 minutes.

These examples indicate the necessity to consider and to minimize input-output time. Operations within a computer occur at phenomenal speeds but accessing peripheral equipment is relatively slow. This is particularly true when random access is sought from sequential tape. It became clear, as the pilot run proceeded, that the use of a random access disc would have saved valuable computer time.

The final product, the audio tape recording is heir to a number of defects and must be carefully checked by the investigator. If the calibration tone output by the program is not used to set the recording level control of the tape recorder, at maximum, the optimum signal to noise ratio is not realized. This results in a background hiss, evident during quiet periods, with possibly perturbing psychoacoustic effects. Poor quality, or previously used imperfectly erased tapes produce the same effect. Even excellent tapes may be degraded, however, by subsequent careless handling by "print through" which can be aggravated by exposure to weak magnetic fields, high temperatures,

and excessively tight winding. The problem is more than theoretical. A number of tapes carried in plastic containers via subway trains, had to be redone because of "print through".

If all audio tape problems are avoided, the output of the digital-analogue converter bears watching. A number of tape recordings had an occasional random click which interfered with the testing and had to be remade. Suspicion centered on the DAC though tape imperfections called "drop outs" may have been the cause.

When all precautions were observed, the production of four hours running time of audio tape, required very much less work than the production of comparable tapes using hardware. The investigator's time was devoted mainly to checking the final tapes while the computer facility personnel were only required to set up the conversion system, start, calibrate and stop the tape recorder.

C. Analysis of Results

Maximum efficiency in the analysis of subjects' responses was not realized in Section 2 since it was not feasible to enter the responses automatically. It became necessary, therefore, to enter 64,000 responses via terminal typewriter, a time consuming process. In

addition, only statistical analysis and display of results were handled by the computer, so that only efficiencies typical of computer usage in general were realized. The potential advantages for analysis of results, accruing from combining the generation programs with automatic recordings of subjects' responses, though they do seem self evident, are yet to be tested.

D. Psychoacoustic Advantages

The primary motivation for the generation program was to supply to psychoacoustic research, a capability beyond that presently available from hardware sources. The investigation into the detection of pure tones masked by noise of varied bandwidth, described in Chapter IV, was designed to test this attempt. The investigation was also undertaken for its own intrinsic interest and seemed to be a particularly appropriate application of the generation program since previous investigations into this area had been hampered by the difficulties associated with hardware noise and analogue filters. The generation program was not only expected to eliminate these difficulties but to provide, in addition, a degree of control over signals which had, previously, been unavailable. It should be borne in mind that the application of the generation program in this study, lies at the zenith of

difficulty for hardware sources but is only a restricted application of the potential inherent in computer generation of signals.

CHAPTER II.

LITERATURE PERTINENT TO THE DETECTION OF TONES IN SUB-CRITICAL BANDWIDTH NOISE

I. Organization

This chapter reviews literature pertinent to the detection of tones masked by noise of sub-critical bandwidth, this being the topic of the investigation to be described in Chapter IV. The review covers four areas: (1) studies determining the width of the critical bands and the detection process within its borders, (2) studies of the effects of the duration of a signal upon its detection, (3) studies measuring the effects of the temporal proximity of gated maskers on a neighboring signal, and (4) studies concerned with the discrimination of changes in signal parameters. These studies are reviewed under the headings, Critical Bands, Temporal Summation, Transient Masking, and Differential Sensitivity.

The presentation plan, for each area, is to first define the concepts, then to discuss its pertinence to the dissertation investigation, and finally, to review the selected studies.

The review is intentionally broad in scope so as to provide an overview of these four psychoacoustic areas which

study detection phenomena.

II. Critical Bands

A. Concepts

The concept of the critical band is credited to Fletcher (1940) who set out to estimate the physical dispersion of exitation along the basilar membrane resulting from noise stimuli. His conclusion, based partially on experimental data but largely on assumption, was that in the presence of a wide band masking noise, the ear acted as though it contained a band limiting filter so that only a relatively narrow band of noise surrounding the signal was effective in masking. This band, he called the "critical band".

In another investigation, Hawkins and Stevens (1950), using one of Fletcher's assumptions namely that the detection threshold for a pure tone masked in noise was reached when noise and tone energy were equal (equal energy hypothesis), calculated the critical bandwidths across the spectrum by experimentally determining the ratio of pure tone pressure to noise pressure per cycle at threshold. This ratio, converted to an energy ratio, is called the "critical ratio" and was used to estimate the width of the critical bands. (The results, however, depend critically upon the assumption of signal to noise energy relationships at threshold.)

Since the effect of the critical band is felt in most areas of interest to psychoacoustics such as loudness and pitch perception, consonance and dissonance, separation of tones, phase perception, etc., the more general definition proposed by Scharf (1970) is appropriate:

> "As a purely empirical phenomenom, the critical band is that bandwidth at which subjective responses rather abruptly begin to change as a band of noise or complex tone increases in bandwidth....".

B. Measurement Techniques

The width of the critical band has been measured in many ways. In addition to the masking effects on pure tones by noise or other tones, subjective changes in loudness, musical consonance, the perception of roughness in monaural and binaural beats, the ability to distinguish AM from FM, and other means, have been used to estimate critical band widths. Since the investigation discussed in the next section is concerned with pure tones masked by noise, the critical band literature reviewed below will deal mainly with measurements based upon masking effect. In this area measurement techniques divide into three; the direct approach, the indirect approach, and mathematical modeling.

The direct approach investigates the critical band by measuring signal thresholds in the presence of masking noise of various bandwidths and, occasionally, band shapes. This

approach tends to make the fewest assumptions and often directs attention to the detection process within the critical band as well as at the border.

The indirect approach accepts some assumption about the detection process, often the equal energy hypothesis, and on that basis seeks to determine the width of the critical band. This approach can reveal very little about subcritical detection, but it is an efficient way to measure the critical band phenomena across frequencies and intensities provided the initial assumption is valid.

The third approach, mathematical modeling, uses the critical band as one of its elements, almost always as a physical filter. Since mathematical modeling compels precise specification of this filter, as well as the entire detection hypothesis, appreciable disciplined speculation is developed on both sub- and supracritical detection.

C. Studies

Fletcher (1940) introduced the critical band concept, but he presented only sparse experimental support for his two major assumptions; (1) the critical band acts as though it were a rectangular band pass filter eliminating all effects of masking noise outside the band, and (2) the detection threshold of a signal in masking noise is reached when the effective energy of the noise equals that

of the signal. These assumptions, however, have exerted a strong influence upon subsequent investigators.

Many investigators, following Fletcher, accepted these assumptions and either used the second assumption to establish the critical bandwidths as a function of frequency (Hawkins and Stevens, 1950), or used both to propose and explore various auditory models containing a band-pass filter to represent the critical band (Swets, Green, and Tanner, 1962), or to explain subjective reactions apparently linked to bandwidth effects (Garner, 1947; Bilger and Hirsh, 1956; Scharf, 1959, 1961, 1962; Plomp and Levelt, 1965.)

Other investigators attempted to directly measure the critical bandwidth either by finding the thresholds of tones masked by variable bandwidth noise (Schafer, Gales, Shewmaker, and Thompson, 1950; Hamilton, 1957; Greenwood, 1961a; den Brink, 1964; Bos and de Boer, 1966) or by masking a probe signal, either tone or narrow band noise, by flanking signals, tone or narrow band noise, adjustable in frequency separation , (Scharf, 1959, 1961, 1962; Green, 1965; Canahl, 1971). In addition other strategies directed toward establishing critical bandwidths were tried, motivated mainly by the pass band filter model (Garner (1948) -- duration effects; Bilger and Hirsh (1956)--linearity of masking vs. masker; Bourbon, Evans, and Deathrage (1968)--

effect of changing noise band skirts; Canahl (1971)--masking linearity; Srinivasan (1971)--signals with similar acoustic qualities but dissimilar spectrums).

Some of these studies supported Fletcher's equal energy hypothesis and his bandwidth estimates as well (Schafer and Gales, 1949; Schafer, Gales, Shewmaker, and Thompson, 1950). Others, either supported, or did not examine the equal energy hypothesis but gave wider estimates of critical bands than did Fletcher (Garner (1949)-- 3 to 4 times wider; Greenwood (1961a, 1961b)-- 2.5 times wider). Still others neither supported Fletcher's equal energy assumption nor his critical bandwidth estimates (Hamilton, 1957; den Brink, 1964, Bos and de Boer, 1966; Green, 1965; Canahl, 1971).

The major source of disquiet about critical band hypotheses does not arise from conflicting bandwidth estimates. De Boer (1962) points out that almost all of the data generated by investigators reporting disparate critical bandwidths, can be made to fit a single curve if adjustments in proportionality constants, or revision of criteria as to where the critical band terminates, are made. The most important unresolved questions concern the subcritical detection mechanism. Fletcher's equal energy assumption when translated into a plot of threshold intensity of signal to noise spectrum level versus noise

bandwidth, generates a straight line, for subcritical bandwidths, with a slope of 3 dB/octave bandwidth. Hamilton (1957) found a slope of 1.5 dB for tone signals. Den Brink (1964) found a non-monotonic curve with a depression just below critical band for tone signals. Bos and de Boer (1966) found a 3 dB slope, within the critical band for both noise and tone signals, a flat supracritical section as per Fletcher for tone signals but only a slight, almost non-significant supra-critical band effect for noise signals. Ahumada and Lovell (1969) also found much wider critical bands for noise signals than for tones. Green's (1965) and Canahl's (1971) data also indicate a subcritical slope less than 3 dB. These discrepancies raise questions about detection within the critical band which are crucial to the validity of the concept itself.

In addition to the above dispute, questions about the critical bandwidth stability versus masking level are, also, unresolved. Fletcher (1940) implied a critical band which is unaffected by masking level. Hawkins and Stevens (1950) and Greenwood (1961a) supported this implication though Greenwood noted sudden changes in critical band-width as levels increased. However Bourbon, Evans, and Deathrage (1968) and Canahl (1971) presented strong evidence indicating a widening of critical band

with increasing masker level.

The sharp nexus at the critical band limits, assumed by Fletcher, implies a rectangular critical pass band. Most direct investigations, however, showed rounded transitions at the borders implying smooth non-rectangular pass bands (Schafer, Gales, Shewmaker, and Thompson, 1950; Bilger and Hirsh, 1956; den Brink, 1964; Bos and de Boer, 1966). A number of attempts have been made to establish the shape of an assumed critical band filter. Schafer, Gales, Shewmaker, and Thompson (1950) proposed the smooth rounded shape generated by a single tuned electronic circuit. Bilger and Hirsh (1956) investigated the skirts of the filter with external filters with two cut-off characteristics. Pfafflin and Matthews (1962) proposed a model comprising a singly-tuned resonant filter followed by a square law detector and an integrator. Swets, Green, and Tanner (1962) estimated the required bandwidths for the assumption of critical band filters of either, single tuned rectangular or Gaussian shapes. Srinivasan (1971) compared two of these possibilities, rectangular versus single tuned and chose the rectangular case as most appropriate for models of human detection.

Considerations of critical band stability, arose,not only with respect to masking level, but also for temporal effects. Hamilton (1957) and den Brink (1964) both noted an apparent widening of the critical band for decreasing tone signal duration. Greenberg (1969) Greenberg & Larkin(1968) and Ahumada & Lovell (1971) investigated a newly

discovered human ability to listen over several critical bands, or a much widened critical band, for signals of unknown frequency and to adjust critical bandwidth center frequency and possibly width, to facilitate the detection of a signal of known frequency.

III. Temporal Summation

A. Concepts and Pertinence

Temporal summation refers to the apparent dependence of human hearing in detection and discrimination tasks, on signal duration. Temporal summation and studies of signal detection in sub-critical bandwidth noise are intertwined since they are both aspects of detection and discrimination. Studies of temporal summation in quiet, detection as a function of duration, often have recourse to critical band theory to explain data. Sub-critical band detection studies of masked thresholds for tones of varying duration and temporal summation studies in masking noise are, clearly, studying the same effect. The duration-sensitive critical band filter which widens as signal duration decreases, admitting more masking noise and, thereby,worsening threshold, represents an alternative formulation for temporal summation. Models of temporal summation effects sometimes include band pass filters and the rationale for some temporal summation investigations start from critical band theory.

B. Studies

Investigators into the effects of signal duration base their efforts on the observed fact that both detection and discrimination improve with increasing signal duration up to some limiting value. Though these studies are, for convenience, subsumed under the label Temporal Summation, the designation is not entirely descriptive of all duration effect hypotheses. This is particularly true when dealing with hypotheses concerning the discrimination of duration increments, or gaps between pulses.

Early workers, and some later ones, devoted their efforts to measuring duration effects and to summarizing them analytically. Hughes (1946) probably provided the earliest equations, suggesting, by implication only, that a summation of energy was occurring. Garner (1949) noted that the detection of pure tones improved linearly with increasing duration between 12 and 200 msec. but that the detection of wide band noise behaved non-linearly within the same interval. Green, Birdsall, and Tanner (1957) divided Hughes equation into three contiguous regions. Miskolscy - Folder (1959) lent support to the invariance of energy summation with envelope and Plomp and Bouman (1959) replaced Green et al's multiregional equations with a single exponential one.

Miller (1948) suggested a physiological explanation of temporal summation by reference to neural transmission times

from peripheral to higher auditory centers. Summation occurs, he asserted, because transmission speeds are direct functions of excitation levels and effects akin to cross and auto-correlation occur. Zwislocki (1960) presented an analytical explanation which posited quanta of stimuli generating exponentially decaying excitations, the summation of all these excitations as a function of time, producing, analytically an exponential summation function. No additional fundamental explanations, excluding some filter detector models, were offered until Abel (1972) proposed that gap detection and duration discrimination, at least, were not dependent upon energy summation but could arise from a simple count of neural firing impulses.

Prior to Abel investigators typically attributed all duration effects to some form of energy summation. Hughes, Miller, Green et al, Miskolsky - Folder, Plomp and Bouman, mentioned above, tacitly, implicitly or explicitly assumed energy summation. Gales and Wilcott (1954), Zwicker and Wright (1963), Dallas and Olsen (1964), Hennings and Psotka (1969) and Campbell and Counter (1969) devoted their efforts to substantiating or refining previous analytical summarizations of duration effects in detection and discrimination. The exponential functions of Plomp and Bouman, and of Zwislocki gained wide acceptance and led to efforts to

establish the time constants of the function. Miskolscy - Folder (1959) and Green, Mckey, and Licklider (1959) found "critical time" limits, primitive time constants from one viewpoint, of approximately 150 msec, beyond which temporal summation does not function. Watson and Gengel (1969) found time constants for Plomp and Bouman's exponential expression but unlike Miskolscy - Folder and Green, Mckey and Licklider, found them to decrease with increasing frequency. Campbell and Counter (1969) supported Watson and Gengel's time constants for middle and higher frequencies but found that temporal summation was inoperative near and below 125 Hz where periodicity pitch begins to appear. (The theory of periodicity pitch holds that at frequencies below 200 Hz, pitch perception results from the time response of neural elements.)

The inadequacy of energy summation as the mechanism underlying all duration perception was exposed when Rochester (1971) proposed an energy summation model to summarize both the detection of pulsed signals and the discrimination of signal duration. His model, though it fitted most of his data satisfactorily, had obvious contradictions which he was forced to note, chiefly that on the basis of his model, or on energy considerations alone, a listener could not discriminate between a very short tone and a very long one of equal energy. Miller and Taylor

(1948) and Garner (1948) anticipated duration discrimination investigations when they measured listeners' ability to discriminate interrupted noises and tones with various repetition rates and duty cycles, from continuous ones. But in this case too, a satisfactory theory was built based upon, a concept related to energy summation, the non linear relationship between intensity and loudness.

Williams and Perrott (1972) continued work in duration discrimination as opposed to detection or intensity discrimination, by investigating the ability to detect gaps between pulses or to resolve two closely spaced pulses. This ability, too, was found to be a function of pulse duration, as well as frequency. Abel (1972) continuing some previous work in gap detection, investigated duration discrimination and proposed the previously mentioned hypothesis modeling this ability by a system which counted neural firings.

Previous to the work by Zwicker and Wright (1963), most studies of temporal summation in masking noise, had used continuous noise. Zwicker and Wright investigating overshoot effects had to use gated noise in order to measure the effects of signal delays relative to noise onset. Interest in the effects of gating masking noise as well as signal, was extended by Campbell (1969) who found that gating both noise and signal simultaneously worsened detection thresholds,

supporting some comments of Tucker, Williams, and Jeffress (1968).

Finally it is worth noting that a number of investigators found it necessary to invoke critical band theory either to rationalize their results or to justify their procedures. Plomp and Bouman (1959) explained the unpredicted increased steepness of the detection versus duration function for very short pulses, by recourse to energy spill outside the critical band and were able to arrive at reasonable estimates of critical band width using this hypothesis. Tucker, Williams, and Jeffress (1968) assuming a band pass filter, energy detector model, looked, though unsuccessfully, for a detectability peak when signal duration became the reciprocal of bandwidth. Leshowitz and Wightman (1971) offered an ambitious explanation of various signal-noise conditions under which temporal summation either functions, or fails, on the basis of a filter-detector model with a filter adjustable in center frequency and positioned by the listener at optimum signal to noise ratio.

IV. Transient Masking

A. Phenomena

Two manifestations of transient masking phenomena are reviewed in the sections following. One, called overshoot refers to a well established empirical observation that the

threshold of a brief signal in the presence of a gated masker is, under some conditions, dependent upon its temporal position in respect to the onset or termination of the masker. When overshoot does occur, the signal threshold always degrades as signal and masker onset approach coincidence. The other manifestation of transient masking, equally well established, is called backward and forward masking and refers to the fact that maskers can degrade the thresholds of signals which precede them, in time, ie, backward masking, as well as those which follow them, ie forward masking. Though overshoot is measured, typically, with the probe signal immersed in the masker while backward-forward masking is measured with signals preceding masker onset or following masker termination, the two are probably very closely related if not aspects of the same phenomenon, overshoot being the forward part of backward-forward masking plus a steady state effect.

B. Pertinence

Transient masking studies are pertinent to sub-critical detection investigations in at least two ways. One, they suggest precautions which must be taken to prevent the intrusion of transient effects into essentially, steady state critical band experiments and two, they introduce data about the auditory system which must be taken into account in formulating hypothesis about the detection of signals in narrow

band noise. However, the precautions suggested by transient masking studies apply mainly to critical band investigations using gated maskers and gated signals. As discussed below, transient masking effects occur for certain combinations of short signals and maskers, when the signal follows, or precedes, the masker relatively closely. These effects, when they occur, can seriously distort the resulting data. Fortunately, however, it appears that delay in signal onset relative to the masker's onset, of 200 msec. or more negate transient masking effects. In addition to onset effects terminal overshoot and adaption must also be considered in experimental designs, such as 2ATFC where one presentation may follow another very closely.

The theoretical pertinence of transient masking to sub-critical band detection lies primarily in the fact that it is a temporal phenomena requiring a mechanism sensitive to time. This kind of mechanism is not unlike those suggested by some investigators to explain duration effects within and across critical bands. This relevancy is illustrated by the adjustable filter hypothesis advanced by some investigators to explain overshoot and by others to explain duration effects upon critical bandwidths. Of equal pertinence is the observation made by several investigators that overshoot occurs on the edges of narrow band stimulation. This effect has, for example, been used to measure critical band width by means of overshoot.

C. Studies

Miller (1947), interested in the masking effects of gated tones, reported significant threshold shift as a function of onset delays between signal and masker. Subsequently other investigators reported this overshoot effect to be present under some conditions and absent under others but it was Zwicker (1965) who first proposed the summarizing hypothesis that the degree of overshoot depended upon the degree of dissimilarity between the signal spectrum and that of the masker. Later, this hypothesis required modification when Elliot (1965) who, in addition to supporting most of Zwicker's findings, also found overshoot to occur for tones at the upper and lower edges of narrow bands of noise. Elliot suggested that overshoot could also be expected in regions of rapid excitation change.

Green (1969) supported Elliot's contention by finding large overshoots for tones lying relatively closely above or below tonal maskers. He advanced two hypothesis to explain these effects, one, a time varying critical band which tuned and narrowed following exitation, or more preferably to him, an energy splatter into the signal region caused by abrupt masker transitions.

Elliot (1969) in an investigation using noise, continuous except for adjustable gaps, showed that both forward and backward masking interacted in the gap periods and

that overshoot plus, probably, all other effects occurred at the same time. Further support was found for her rapid excitation region and terminal overshoot was also found. Leshowitz and Cudahy (1972) also found large overshoots present at the terminal transition of the masker as well as initially, and attempted to implement Green's (1969) suggestion about energy splatter using short term fourier transforms. Unfortunately Leshowitz and Cudahy do not show how overshoot functions arise analytically from the transforms.

Meanwhile the overshoot phenomena had been found to exist for contralateral masking. This effect, called transient central masking was investigated by Zwicker, Buining, and Glantz (1968) who in addition to measuring overshoot to 15 dB dichotically, were able to determine frequency distributions of masking, obtaining bandwidths not inconsistent with those obtained by critical band investigators for moderate levels. However, unlike most reported results, they found bandwidths to increase with masker level.

The very close relationship between overshoot and forward-backward masking is emphasized by the fact that Miller (1947) also noted backward and forward masking effects in his work with gated noise. Pickett (1959) measured these effects for various tone durations, silent intervals between

tone and noise, and noise levels. He found appreciable threshold elevation for silent intervals less than 25 msec. long. Elliot (1962), measured, forward-backward masking over intervals to 100 msec. first at 1000 Hz and later at several frequencies (Elliot, 1926), for dichotic and monotic conditions. She found forward-backward masking to extend to approximately 100 msec. and some evidence of slopes dependent on frequency.

Closely allied to temporal masking is the perception of temporal order which Hirsh (1959) investigated. He found that, for a large variety of acoustic qualities, a separation of 17 msec., approximately, was required for correct perception of order. Homicke, Elfner and Bothe (1969) extended Hirsh's work, concluding that the detection of probe tones in forward-backward masking and the perception of temporal order had similar underlying mechanisms since both phenomena responded to changes in signal level and to changes of signal position relative to the noise band, in similar ways.

In summation, it should be noted that transient masking studies are much stronger in terms of empirical data than in unifying hypothesis. As Elliot concluded, these effects are more easily described than explained.

V. Differential Sensitivity

A. Definition

Differential sensitivity is the term applied to the ability of the auditory system to discriminate changes in signals. These changes may be in frequency, rate of change of envelope, diminution, intensity or others. Interest in this section lies mainly in intensity change discrimination but diminution and rate of envelope change are also considered.

Though, initially, discrimination of intensity change had been viewed as distinct from the detection of a signal masked by a background signal, the essential unity of the two is now widely understood. The discrimination of increments in pure tones constitutes, in fact, the lower end of the continuum arising from the experimental paradigm for detecting the presence of a pure tone in noise of continually narrowing bandwidth. At very narrow bandwidths, the noise becomes indistinguishable from the signal generated by a pure tone oscillator and the task of detecting a pure tone in noise becomes that of discriminating an increment in a pure tone. The discrimination of changes in other kinds of signals, noise, complex tones, etc., though of equal import are, however, not contained in this continuum.

Though the two tasks, detection and discrimination, may bracket a continuum, discrimination in quiet, does confront the ear with situations which differ from detection. These

differences devolve from the fact that in discrimination the signals upon which decisions are to be made are always clearly perceived and known while in detection the signal to be detected is only remembered. If memory or foreknowledge of the signal plays a role in perception, this factor should operate more strongly in the first instance. The interaction of this factor, if indeed present, with the energy detection mechanism assumed to operate in detection, is one of the reasons that the study of differential sensitivity is so pertinent to the study of critical bands. It should be clear, also, that people are more often called upon to perceive changes in acoustic signals, speech for example, than simply to detect the presence of a sound. Discrimination is, therefore, a more important task to the auditory system than detection.

Differential sensitivity has been expressed by several kinds of units, JND, JNND, DL, and IU. The JND (just noticeable difference) represents the average of the smallest increments perceived. Since decrements can also be measured the JNND (just noticeable negative difference) has also been used. The JND or JNND expressed as a ratio referred to the base signal parameter, is often called the DL (difference limen) (Littler, 1965). Some investigators noting that the subject may respond either to the increment or decrement or both, despite the intention of the inves-

tigator, have made use of the term IU (interval of uncertainty) representing the difference between (the JND and the JNND) which are just perceivable by the subject. Though the units discussed above generally refer to intensity and therefore contain the assumption that the sensitivity to changes in signal intensity is a function only of intensity, it should be noted that this concept has often been challenged, other factors such as rate of change of envelope being considered significant, as well.

B. Pertinence

The immediately preceding section has already implied that, for pure tones, the DL is a measure of the signal to noise ratio at threshold for the tone centered within a very narrow band of noise. It should be expected, therefore, that curves of subcritical detection would, at their narrowest noise bands, anchor at established DL values, after adjustment for possible phase differences between the pure tone and the noise. This requirement for models of sub-critical detection has not always been recognized but it appears to be justified.

Differential sensitivity studies also reveal the auditory system dealing with signals and maskers without qualitative difference. At the opposite pole of the continuum, supra-critical detection studies of pure tone in wide band

noise reveal the same system dealing with a signal differing qualitatively from the masker. Though some investigators ignore the role of qualitative differences in detection, common experience makes this position appear somewhat dubious. Since both detection conditions must be taken into account, differential sensitivity studies are obviously highly pertinent to sub-critical detection theories.

C. Studies

Knudsen (1923) did early work on differential sensitivity but his pre-electronic techniques were considered prone to introduce transients. Riesz (1928) conducted one of the earliest studies using electronic techniques. He used the beats produced by two closely tuned oscillators to generate tonal increments with smooth, gradual transitions. Riesz's results indicated the JND to be a function of both frequency and level. Riesz had worked only with pure tones but Miller (1947) investigated the JND for wide band noise. Though his JNDs decreased more rapidly with increasing level than did Riesz's, he, nevertheless, found the same trend and the same higher level magnitudes. He concluded, consequently, that the auditory mechanism for noise discrimination was similar to that for pure tones.

Other investigators, Small, Bacon, and Fozard (1959) noting that listeners could, unknowingly, be responding to

intensity decrements as well as increments, introduced the concept of the interval of uncertainly (IU). Their data supported Riesz's results in trend but deviated slightly in magnitudes.

A number of more recent investigators appear, however, to disagree with both Riesz and Miller. Small and Minifie (1961) measuring the JND of a tone in the presence of noise, found it to be a function of signal to noise level rather than solely of level as reported by Riesz. Their insistence that this result contradicted Riesz is not clearly justified since his experiment did not include masking noise. Harris (1963) in a test in which the subject controlled the number of exposures to the test prior to making a decision, achieved results apparently indicating that the DL was invariant with loudness and frequency. A second test, however, using the method of adjustments, not only gave larger DLs but showed greater loudness dependency for the 1000 Hz signal. Loeb and Binford (1963) noting as did Harris that the DL was task sensitive, compared DLs for continuous tones, occasionally incremented, to those for similarly incremented gated wide band noise. They found that the continuous condition gave lower DLs than the gated condition and was also less degraded by fatigue. Wolsk (1964) asserted that Riesz's work contained artifacts arising from his lack of control of rate of change of

envelope. He performed experiments in which this factor was controlled and found contradictions to an experiment by Riesz which showed that warbling rate did not effect the DL.

Meanwhile some investigators had focussed on an essential difference between the detection of a pure tone in wide band noise and tone discrimination, the first condition obeying Weber's law, the second, not. Tanner (1961) noted that analysis based upon signal detection theory predicted a deviation of tone discrimination from Weber's law but not of noise discrimination. His experimental work, however, showed no such deviation. McGill and Goldberg (1968) did find empirical evidence of the failure of Weber's law for tone discrimination and proposed a model based on neural firings. Viemeister (1972) also noted this deviation as being well established and advanced the hypothesis, supported subsequently by tests, that listeners made use of intensive information lying above the frequency of the signal.

A number of hypotheses and models have been proposed to explain differential sensitivity to intensity. Miller (1947) suggested neural excitation in the presence of fluctuations in listeners sensitivity. Tanner (1961) applied a filter integrator detector model to discrimination.

Creelman (1961) emphasized the role of memory, suggesting that listener acted similarly to a signal-known-exactly detector but with faulty memory. Loeb and Binford (1963) hypothesized that the subject listened at intervals in which he was physiologically ready and showed experimentally, as was predicted by the hypothesis that gating the signal raised the DL. McGill and Goldberg (1968), as previously noted, proposed a model in which stimuli were transformed to trains of pulses, discrimination decisions being based on the pulse count. Hennings (1968) noted the failure of most models of discrimination at higher levels where they typically predict error free discrimination. He found two models acceptable, one by Green and Swets (1967) which added sources of extraneous noise, and his own which accomplished the introduction of uncertainty and the degrading of high level discrimination, by using a time-jittering filter. Viemeister (1972) developed a model based on the perception of aural harmonics.

In addition to the work in differential sensitivity to intensity, studies have been pursued for other signal parameters. Kirk (1958) investigated sensitivity to changes in the diminution rate of pure and complex tones. Creelman (1961) investigated the effects of decay rates on damped

sinusoids for both detection and discrimination. Wolsk (1964) found that the intensity DL decreased with increased rate of envelope change for 2 out of 4 subjects but did not support the warble rate effect upon DL reported by Riesz.

Investigations have also been made into the ability of humans to discriminate small changes in frequency. These studies are pertinent since they illustrate extremely sensitive modes of detection. Shower and Biddulph (1931), for example, found the discrimination of frequency changes extended down to DL's of -27 dB. Harris (1952) using a different technique found similar results. The difference limen, for frequency, according to both investigators, appears to decrease both with increasing frequency and level of base frequency but disputes do exist about the nature of this decrease particularly at low frequencies.

CHAPTER III.

PRELIMINARIES TO THE MAIN INVESTIGATION

I. Introduction

This section comprising Chapters III and IV, describes an investigation into the detection of pure tone pulses masked by Gaussian noise of variable bandwidth. The test signals, both noise and tones, were generated entirely by computer using the programs described in Chapter I. The masking noises vary in bandwidth by octaves across the range 1.25 Hz to 640 Hz. This range reflects an intent to encompass the continuum from masking by wide band noise to masking by pure tones.

Because of an interest in the possible effects of specific features of the stimulus as well as the traditional parameter of noise bandwidth, a single reproducible noise was used throughout each run and two different reproducible noises were generated per noise-bandwidth specification. In order to eliminate the objections and doubts to be raised below in regards to previous investigations, the parameters of the generated signals were precisely controlled.

The motivation for this investigation arises from the difficulties encountered by other investigators in their

attempts to produce the required narrow band noises using hardware sources and filters and, also, by their enforced dependence upon imprecisely controlled equipment for all noises. The first difficulty has led either to limitations on the lower bounds of the noise bandwidths so that the entire continuum could not be investigated; to the use of troublesome, not completely satisfactory analogue filters; or to the adoption of incompletely justified assumptions about the nature of noise. The second difficulty has led to the acceptance of the introduction of an extraneous variability into the data--the presentation to presentation intensity mean variation of the noise source--and a consequent, but doubtful reliance upon statistical averaging to reduce this effect.

At present, as a result of these difficulties, there are a number of important, unresolved questions about the nature of sub-critical detection. These questions concern detection mechanisms invoked by listeners in the presence of noise which is perceived in qualitatively varied ways as its bandwidth changes.

A. Quality Difference

The quality difference between a tone and noise, a factor typically ignored, diminishes as the bandwidth of an

initially wide band noise decreases. In the course of this progression, the perception of the noise changes from a hissing sound, through an atonal sound of increasing roughness, into a region in which a fluctuating tone of decreasing rate is heard, and finally, to a sound indistinguishable from a pure tone. It is difficult to accept the assumption that these qualitatively different perceptions play no role in detection, that the listener makes no use of these distinctions to ease the difficult detection task. If, as seems more likely, he does make use of this information, then he is really being presented with a series of complex and sophisticated tasks, in the course of a tone in variable bandwidth noise investigation, beyond either the knowledge or the intent of the investigator.

A number of investigators have perceived these possibilities. Licklider, Marill, and Neiser (1954) labeled the quality difference between a tone and wider band noises, as salience and Hamilton (1957) attributed his unexpectedly flat 1.5 dB per octave bandwidth sub-critical slope to this effect. Bos and de Boer (1966) suggested that the critical band effect, rather than resulting from internal filters, emerges because of the increasing detectability of tones against widening bandwidth noise.

B. Noise Fluctuations

A number of investigators have also suggested that detection, within the region in which the noise is perceived as a tone fluctuating at rates related to bandwidths, may be degraded by the signal-like changes in the noise envelope which mislead the listener or force him to raise his criteria. Pfafflin and Matthews (1966) noted that bias factors arising from stimulus features unconnected with the presence of the signal, affect detection. Den Brink (1964) referred to the effect of these envelope fluctuations upon the subject's criteria as a criteria effect and, to it, accounted the disparity between his results and Fletcher's (1940). Bos and de Boer (1966) also concluded that the subcritical threshold was determined primarily by envelope fluctuations and, to these fluctuations, Canahl (1971) attributed the 10 dB per decade change in detectability of a tone surrounded by two masking noises which changed spectral positions symmetrically.

C. Pattern Recognition

For reproducible noise (frozen noise), an additional potential detection mechanism exists, namely, pattern distinction which may be feasible for the listener when the noise pattern is remembered. This mechanism may yield large improvements in thresholds for some subjects for some noises. Pfafflin and Matthews (1966) using computer generated frozen

noises, reported near perfect recognition rates for some subjects for some noises. Pfafflin (1968) also found that the subjects recall of noise stimuli acted to improve detection and that this effect was, occasionally, very pronounced.

D. Opposing Views

There exists, at present, opposing views about the nature of sub-critical detection based upon apparently contradictory data. With some violence, these views may be forced into two categories; one exemplified by Fletcher (1940) holding that sub-critical detection depended only upon signal to noise energy ratios, the other by den Brink (1964) who found that additional factors, related to specific features of the stimulus, influence detection.

Data representative of the two views is displayed in Fig. 21 in a manner which has become traditional. The ordinate, of the display, shows the threshold level in terms of the dB ratio between signal intensity and the spectrum level of the noise. The abscissa shows the noise bandwidth transformed to a log scale so that each doubling of distance along the abscissa represents a doubling of noise bandwidth.

In this representation, Fletcher's assumption that threshold occurs when the signal and noise energies (or

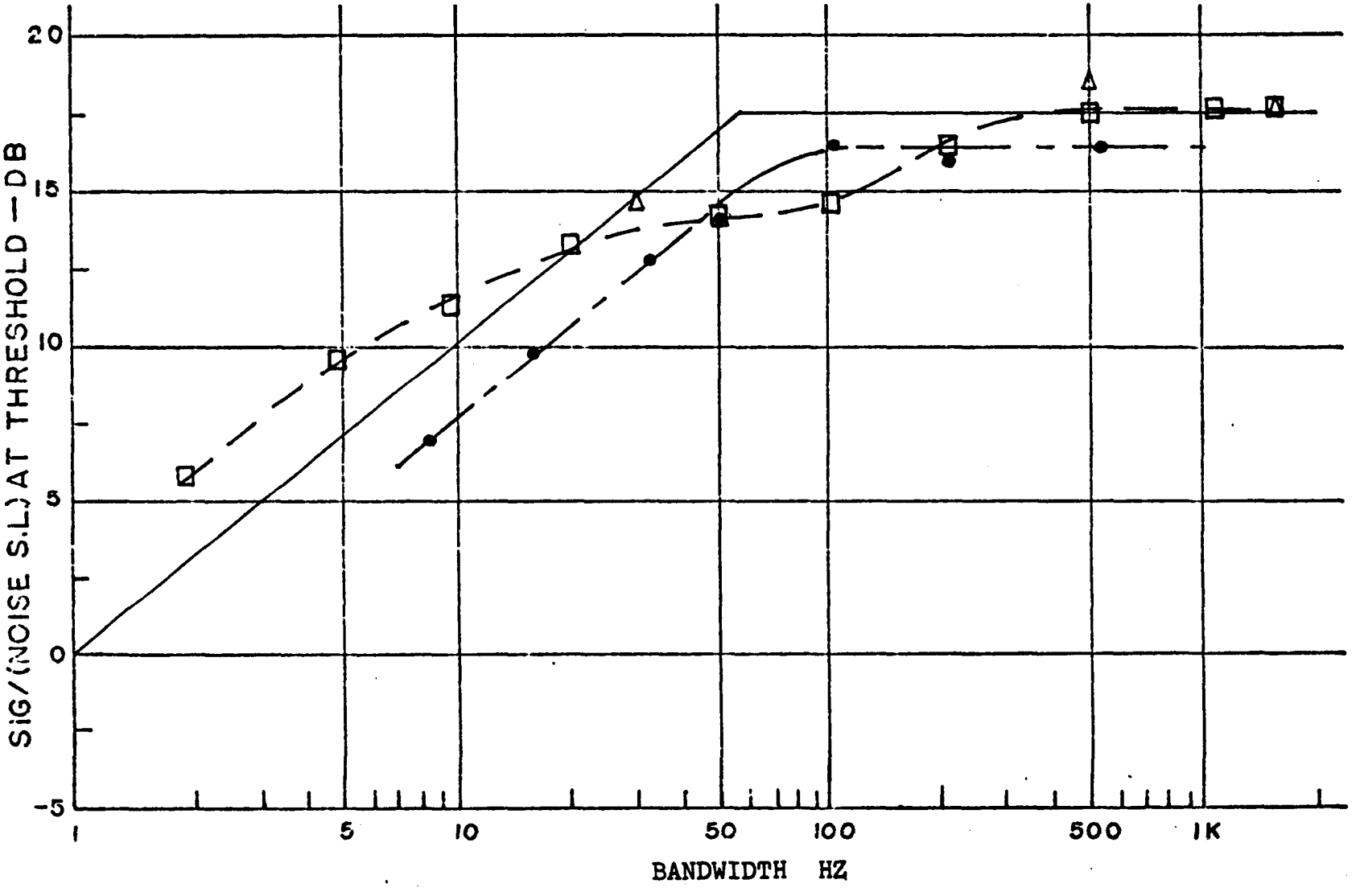
intensities) are equal, generates a straight line originating at 0 db, for 1 Hz bandwidth noise, and rising with a slope of 3 dB per octave bandwidth for all sub-critical noise bands. This occurs because each bandwidth increase requires an equal increase in the signal to noise spectrum level ratio. Above the sub-critical region, increasing noise bandwidth produces no increase in effective masking energy so that the threshold remains fixed. Though this type of display has been widely used, data are customarily taken at constant noise intensity levels rather than at constant noise spectrum levels as implied by the display. This customary procedure has been followed in the pilot test and in the main investigation of this study.

Figure 21 reveals the sparseness of Fletcher's sub-critical data. It shows also, that the data of Schafer, Gales, Shewmaker, and Thompson (1950) support Fletcher in terms of slope but not in absolute values of thresholds, critical bandwidths, or in the abruptness of the sub-critical to supra-critical transition. The data of den Brink (1964), into which Hamilton's (1957) data may be subsumed, are seen to depart significantly from the straight line generated by Fletcher's assumption and imply that factors, other than those related directly to energy or intensity ratios, are affecting detection.

Because both categories are supported by data based on

FIGURE 21: Ratio of Signal Intensity to Noise Spectrum Level in dB versus Noise Bandwidth--

for Fletcher (1940)--"Δ" and solid line; Schafer, Gales, Shewmaker, and Thompson (1950)--"o" and long-short dashed line; and den Brink (1964) "□" and equal segment dashed line.


SIG/NOISE S.L. AT THRESHOLD — DB
20
15
10
5
0
-5
1
5
10
50
100
500
1K
BANDWIDTH HZ

hardware generated signals and because these data diverge most in the narrower noise bandwidth regions where hardware generated noise and analogue filters become most questionable and troublesome, the underlying assumptions and the procedures of these investigations merit scrutiny.

II. Problems of Experiments Using Hardware Generated Noise and Analog Filters

A. Variability of Levels Between Presentations

All users of hardware generated noise sources are constrained, in practice, to control only those long term statistical averages of the noise provided for by the equipment design. Not only is there an inability to control specific features of the noise, but there is an attendant inability to control the statistical averages of individual presentation as well. This means that uncontrollable variations exist in the mean intensity of the stimulus in individual presentations and in the configuration of the presentation envelope. In order to proceed, the investigator using such test signals, must assume that the effects of these variations are averaged out or cancelled by repeated sampling, yielding results which are independent of these variations. In effect, in tests in which such variations are uncontrolled, the investigator assumes that all tests using noises of the same bandwidth and equal overall mean intensities, should yield similar results regardless of the differences which may exist in intra-test variability of the presentation mean.

This assumption cannot be established by statistical analysis

of the test signal, since the assumption concerns the effects of such signals upon the listener. In analytical terms this assumption requires two yet unestablished corollaries; (1) that the effects of the test signals exhibit stationarity which requires that no learning or changing criteria occurs on the part of the subject, and (2) that linearity of effect occurs so that the effect of presentation mean intensities greater than the overall mean are effectively cancelled by those below the mean. For the purposes of the present discussion, the doubtfulness of the assumption, and its corollaries, is seen most easily in terms of specific features of the presentations.

A number of studies, Watson (1969), Pfafflin and Matthews (1966), and the investigation described in this chapter, indicate that subjects respond to specific features of the noise presentation such as voltage peaks, signal-like changes in the noise envelope, and localized variations in average levels in the region of signal onset. Den Brink (1964), Bos and de Boer (1966), and Canahl (1971) suggest that these noise envelope fluctuations confuse the subject and cause him to raise his criteria. Since the variation of the presentation mean about the overall mean is a statistical function of the number of independent samples in the presentation (this number being a function of noise bandwidth and presentation duration), this characteristic may vary with the investigative procedure, consequently affecting the criteria shift and significantly affecting thresholds. In addition, the assumption that the effects of signal-like excursions of the noise envelope are cancelled by other local parameter deviations,

is without support.

Pfafflin and Matthews (1966) present some illustration of the effect of noise variability upon subjects. They compared the results of 2ATFC detection tests for a tone in noise, for two cases; one in which identical noises comprised each member of the pair of presentations; the other, in which the noises were not identical. A plot of individual results showed that in addition to increased variability for the non-identical case, some of its observed proportions were significantly below chance levels even for relatively high signal to noise ratios. As a result, the two cases do not yield similar psychometric curves. This suggests that effects additional to the loss of resolution arise from increased variability of the stimuli.

B. Effects of Starting Phase Variability

One of the stimulus features uncontrolled in the experiments of Fletcher (1940), Schafer et al (1950), Hamilton (1957), den Brink (1964), and Bos and de Boer (1966), is the phase of the pure tone signal relative to the noise, at signal onset. Pfafflin and Matthews (1966) showed that this relationship (starting phase) affected detection in a manner consistent with energy for a 1250 Hz bandwidth noise. For very narrow bandwidths, the effects of starting phase become more striking since the starting phase persists and affects the energy change resulting from the signal addition very strongly and in a manner which is not linearly related to phase variations.

For example, the addition of an equal amplitude (pressure) tone to a very narrow band of noise centered about the tone frequency,

may result in total signal amplitudes varying from twice the noise amplitude to a level close to zero as the starting phase varies from "0" to "180" degrees. In terms of intensity in dB, this is a variation from +6 dB to a very large negative value. While it may be assumed that the random starting phases cluster symmetrically about "0" degrees phase so that the overall mean phase is indeed "0", it is not safe to make the same assumption about the effects upon threshold of such phase shifts since no strong support for this assumption can be found in the literature.

An additional phase effect could arise from the fact that the detection of a pure tone in very narrow band noise is similar to tone in tone discrimination. Since it is generally accepted that the discrimination of tonal increments, or decrements, is a function of the resting intensity of the base tone, wide, though symmetrical deviations of the presentation mean which average out statistically, could nevertheless bias the results via this avenue.

In view of these comments, uncontrolled variations in the starting phase between noise and signal, appear to have the potential to significantly contaminate results.

C. Problems of Narrow Band Filters

A second restraint imposed upon investigators using hardware, arises from the difficulties associated with implementing band pass filters with large center frequency to bandwidth ratios (high "Q"). Schafer et al (1950) attempted to avoid this problem by synthesizing narrow noise bands; Hamilton (1957) avoided these difficulties by limiting his narrowest bandwidth to 19 Hz, but den Brink (1964) met the

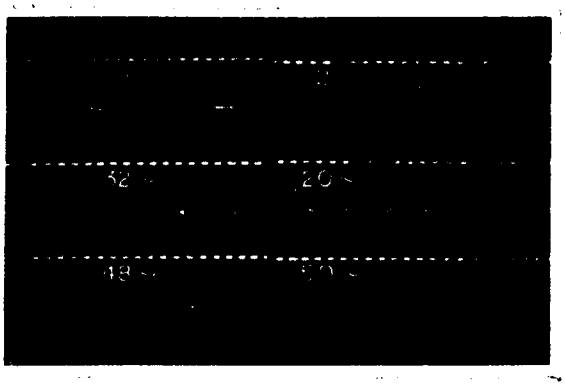
problem directly by using unusually high "Q" filters. Bos and de Boer (1966) attempted to produce steep sided narrow band noise via heterodyning. In all cases questions about these procedures, arise involving transients and envelope distortions.

Fletcher's (1940) investigation of sub-critical detection was largely assumptive and, since no measurements were taken below noise bandwidths of 30 Hz, entailed no difficult filtering problems. Schafer et al, however, in their attempt to test Fletcher's assumptions empirically, did face such problems and sought to avoid them by synthesizing their noises from multiples of groups of 8 tones spaced 1 Hz apart. These synthesized noises are occasionally interspersed with thermal noises of wider bandwidths for comparison. Unfortunately these comparisons are not available in the most crucial region, the narrow band region, so that questions remain about the validity of these synthetic noises. This is so despite the fact that they were compared to thermal noise via listening tests, visual inspections, and measured auto-correlation functions.

A visual comparison of the envelopes of synthetic and thermal noise, indeed, indicates a failure of equivalence as the number of tones comprising the synthetic noise diminish. (See figure 22.) The 8 Hz synthetic noise, for example, appears equivalent to the 2 Hz thermal noise, the 32 Hz synthetic noise appears equivalent to the 20 Hz thermal noise, and have been so juxtaposed by Schafer et al.

This is a serious matter. If the tone groups generate envelopes appropriate to narrower thermal noise bandwidths, as they appear to do,

FIGURE 22: Comparison of Synthetic and Thermal Noises used by Schafer, Gales, Shewmaker, and Thompson (1950).



then the sub-critical slope based upon the equivalent thermal noise bandwidths, becomes, approximately 2 dB/octave instead of 3. In this case Fletcher's assumption is not supported but Hamilton's subsequent estimate of a slope of 1 dB/octave, is.

The probability density distribution of noise must also be considered. Though the probability density distribution of the instantaneous values of thermal noise are known to be Gaussian, the distribution of the synthetic noise is not known. Theoretically, four or more constant amplitude uniformly random phase sinusoids combine to form a wave with an approximately Rayleigh envelope distribution and a Gaussian distribution of instantaneous values. However, though the Gaussian distribution may be approximated for the 8 tone group there is more doubt about multiples of this group. From the viewpoint of specific stimulus features of the noise presentation, these doubts translate into the possibility that the synthetic noises exhibited peculiar envelope configurations which affected detection in some way, possibly including periodicities.

Hamilton, den Brink, and Bos and de Boer used analogue filters to control their noise bandwidths down to bandwidths of 19, 2, and 5 Hz respectively. Unfortunately so little information is given about the various filter characteristics that it is not possible to evaluate their effects upon the results. In no case did the investigators describe the phase characteristics of their filters which, in addition to the magnitude of the spectral attenuation function, is required for a complete description of the effect of filtering upon signals.

This indicates a lack of concern for potential filter distortions or reflects the tacit assumption that the spectral characteristics and long term intensity ratios of signals, are the only significant factors. This assumption diminishes the chances of observing other factors, especially those arising from specific features of the envelopes of the stimuli.

Traditionally filters are described by the magnitude of the attenuation as a function of frequency and by the phase shift across the filter, also as a function of frequency. These are steady state responses and apply to signals which have been passing through the filter for relatively long times. An alternative description for linear systems integrable over long time intervals, is the reaction of the filter to a pulse or an abrupt change of level, ie the impulse or transient response. The steady state describers are most useful for signals of relatively long duration which change levels slowly. For short signals, or those which change levels abruptly, a description of the transient response is most useful.

Since the largest changes in signal level, for both noise and pure tone, occur when the signals are turned on or off, the filter's reaction to these initial and terminal transitions can be an important consideration. The investigators discussed above, have attempted to minimize these transition effects upon the noise signal, at least, by using either continuous noise--den Brink and Bos and de Boer-- or relatively long noises--Schafer et al, 5 seconds, and Hamilton, 2 seconds. Of the four, only Hamilton used the forced choice procedure which is generally conceded to provide more stable criteria than yes-no tests.

Since, however, he does not carry his investigation below noise bandwidths of 19 Hz, he experiences little trouble from the transients caused by the initial and terminal transitions of his noise pulses. In his case these transients are probably no longer than 18 msec. and have subsided by signal onset. If, however, he had attempted to extend his investigation to 2 Hz using 2 second long noise pulses, he would have faced serious difficulties arising from the duration of the initial and final transients.

A more serious question is raised by the fact that Hamilton and den Brink, also passed their pure tone bursts through the same filters as the noise. While their longest pulses probably undergo some initial and terminal distortions, the shortest pulses, after passage through the narrowest filters, may have suffered such severe envelope distortion that they bear little resemblance to the envelope of the input noise.

Den Brink compensates, approximately, for the energy lost by the narrower pulses in the narrower bandwidth filters, but it is not possible to compensate for the distortions of envelope, duration, and other stimulus feature.

Bos and de Boer (1966) illustrate another difficulty arising from a lack of consideration of the effects of filter processing upon signal envelopes. They attempted to maximize the steepness of their noise spectral function by first filtering the white noise at a low center frequency and then heterodyning the output to the desired center frequency. In this process they used a carrier tone, modulated

by noise, a detector and a single sideband filter. This generates a single sideband signal at the desired center frequency, with potentially high "Qs", and with the desired magnitude versus frequency characteristic.

However, the phase-frequency characteristics of such a signal may differ considerably from that produced by filtering white noise directly. Since the envelope of a filtered noise depends upon both magnitude and phase characteristics of the noise spectrum, the resulting noises produced by this technique may differ crucially from those used by other investigators.

One of the results of single side band processing, for example, is to produce a signal with transition rates dependent upon the signal level (Uitjens, 1953). This signal acts as though it had twice the actual bandwidth for small modulations but for large modulations its response is limited by its actual bandwidth. In addition, signals generated by single sideband processing show better resolution than expected (Uitjens 1953). The effects of these, untypical, characteristics upon the signal envelopes is unexplored and unaccounted for by Bos and de Boer.

III. Elimination of Stimulus Variability and Filter Restraints by the Computer Generation Program

A. Introduction

The computer generation program described in Chapter I, permits the main investigation to proceed without the need for doubtful assumptions about the nature of noise and to escape the restraints of

hardware filters. Specifically, the generation program permits the elimination of presentation variability and the minimization of filter distortions. Also provided, is the capability to recreate each noise presentation and to conduct a search for specific stimulus features related to detection. The use of this program permits a finer and less questionable resolution of the nature of sub-critical detection than was possible heretofore.

B. Presentation Level

For reasons given above, the mean intensity of the individual presentation was adjusted to equal the overall mean. Though the program, with slight modification allows this to be done for continuously changing noise, the operation was simplified by the decision to use two frozen noises per bandwidth, a total of 20 noises in this case. A two pass technique was used in which the appropriate multiplier was found for each noise so as to adjust its presentation mean intensity to that of the overall value. Figures 50 to 69 show the final products prior to rise and fall shaping each point on the display representing the RMS value of 100 cycles or 10 msec. of noise.

C. Phase

The phase of the signal relative to the noise at onset, was a parameter of the signal judged to warrant control. The phase was controlled by displacing the onset time of the signal until the first maximum of the signal coincided with a positive slope, zero value noise. This was checked by observing the first twenty or thirty cycles of the combined output of equal intensity noise and signal.

For the narrowest noise bands, this onset phase relationship persists to some extent so that this is equivalent to adding both signals approximately in phase quadrature. For wider band noises, this initial phase does not persist and the signal phase is uncorrelated to the zero crossings of the noise. For all cases, however, the initial phase relationship guaranteed that the addition of a signal to the noise would always produce an initial increment in overall intensity.

D. Filters

The restraints upon filter implementation, discussed above do not exist for filters simulated by computer. The limits on "Q" in this case arise only from the limitation on computational precision which also determines the level of unwanted truncation noise. This program used the double precision mode for all real number calculations, effectively 56 bits. It was shown in Chapter I that the limit on "Q" lay far above the requirements of this investigation and that the truncation noise for the worst case, was insignificant.

The filter algorithm is the impulse invariant digital equivalent of the optimally flat Butterworth quadruple. This means that its transient response is similar to that of the analogue version. This filter gives a good transient response responding to an abrupt increase of input (a step) with an output which rises with a rise time given by

$$\tau = .78/B$$

where "τ" is the rise time from the 10% to 90% of steady state level and "B" is the bandwidths of the filter measured between 3 dB points.

A small initial overshoot is also introduced which rises 10.9% above the steady state value (Uitjens, 1953). A highly selective pass band is achieved (figures 11 and 12); the attenuation of signals 1 and 1.5 bandwidths removed from center frequency being approximately 40 dB below that of center frequency. The overall effect of this filter configuration is to produce a reasonably selective filtered version of the input noise pulse with a minimum of envelope distortion.

E. Envelopes

The dependence of the minimum initial and final transition rise or fall times, and of the minimum duration of noise pulses upon the processing filter, is eliminated by the generation program. This is done by suppressing the initial output of the filter long enough to insure that all transients have subsided. When the filter output has reached the steady state condition, it is passed on for envelope shaping. Consequently, transitions of any shape and duration and overall pulse durations of any length, may be imposed upon the filtered noise without regard to the limitations of the filter's transient response. Transitions are introduced by multiplying initial and final sections of the pulse by the desired expressions and overall duration is controlled by specifying the initial and final cycle number at which the noise is passed on or turned off.

The two noises used here were given post-filtering delays of 16,000 and 20,000 computational cycles, respectively, corresponding to temporal delays of 1.5 and 2 seconds. Since the narrowest filter has an inherent rise time of approximately .60 seconds a post filtering

delay of 1.5 seconds or 2.5 time constants, is sufficient to guarantee the absence of transients from the utilized output of this filter.

F. Stimulus Features

The decision to use one frozen noise per run was made in order to encourage subjects to function at maximum sensitivity and repeatability and to benefit from waveform memory when possible. A pilot run, in which the various bandwidths were mixed randomly proved difficult for subjects and generated relatively poor and widely varying thresholds. Pfafflin (1968) had reported that the use of a single noise per run tended to improve detection generally and that the improvement, morever, was very great for some subjects for some noises. Since these effects, particularly the latter, might relate to stimulus features other than overall intensities, the decision was made to follow Pfafflin's example.

The decision to record two runs per bandwidth, each run using a different frozen noise, was made in the belief that this would reveal the effects of specific stimulus features.

IV. Pilot Test

A. Reason for Test

A practical difficulty encountered in this study concerned the effects of interposing an intermediate stage in the investigation, in which the output of the digital to analogue converter was recorded and stored on analogue audio tape prior to testing. In installations having dedicated computers this stage, with its potential contaminations,

may be avoided. However, since this arrangement was not available, the effects of recording and playback had to be faced.

In Chapter I, the distortions and other extraneous effects introduced into the signal by tape recorders, were discussed. They include two kinds of effects. One is the introduction of a wide band noise with both additive and modulated components arising from tape coating variations, bias current imbalance, tape bounce, and amplifier noise. The second effect originates in the variations in the traversing speed of the tape which produces effects confined, typically, to the 10 to 200 Hz band and called wow and flutter. Though average values may be used to describe these effects, .2% RMS deviations for high grade tape recorders for example, the rate and the extent of the deviations over the short periods of time in which the presentations of the main investigation occur, is not predictable.

The first kind of effect, the introduction of wide band noise, was judged insignificant for two reasons. First, on the basis of measurements made during quiet periods between noises in the main investigation, the extraneous noise was always more than 40 dB below the masking noise (a low pass filter cutting off at 2000 Hz was introduced into the earphone circuit). At its worst, then, it contaminated the masking noise with a continuous noise of unknown statistical distribution, always less than 1% its intensity.

The second kind of effect, erratic and unpredictable changes in the speed at which the tape travels past the recording and playback heads, results from tape slippage, stretch and variations in the

rotational velocity of the transport mechanism. The major effects of these variations is to produce changes in the center frequency of the noise and in the pure tone frequency of about 2 Hz on the average but probably reaching higher values on occasion for the tape recorders used in the main investigation.

It is difficult to judge, apriori, the effect of these variations. Even a 2 Hz change within a noise presentation with a bandwidth of 1.25 Hz, seems extremely severe. However, since this speed change occurs slowly in relation to the duration of the presentation, only a percentage of this change can occur during a presentation. The major effect of this change, then, would be to slightly shift the frequency of the tone and the center of the noise band. This effect would be slight since the spectral position of the tone, at the center of the noise band, would be unaffected and the detection threshold changes slowly with frequency. In addition, an increase of tape recorder speed of even 1% would only widen the noise band by the same amount, an insignificant increase having only tiny effect upon the envelope of the noise waveform. This would be true, even for slow changes occurring during the recording or playback of a presentation.

However since all this discussion is unestablished speculation, the effects, if any, of interposing a tape recorder had to be established empirically.

B. Procedures

Since variations in tape transport speed cause the same small

uniform percentage changes in bandwidth for all noise bandwidths but a constant deviation in the signal frequency and the center frequency of noise bandwidths, the presumption was made that the effects would be most noticeable at the narrowest bandwidth. Consequently, the pilot tests were conducted at the narrowest bandwidth obtainable, the pulsed output of a pure tone generator, to which a pure tone of the same frequency but in phase quadrature, was appropriately added to represent the signal. Since the 1.25 Hz bandwidth noise the narrowest bandwidth is close to being a pure tone, the pilot test may be considered to have been conducted at the narrowest bandwidth of the main investigation.

The test procedure used the two alternative forced choice paradigm with the temporal organization given in Table 6. Except for the shape of the initial and final transitions of the signals which had to be exponential because of equipment and the longer quiet period between trials, the temporal organization duplicated that of the main investigation.

To administer the direct test, the tester sat in view of the subject within a sound proofed room, and carried out the adjustments required above. Prior to the testing, the direct test had been recorded on another tape. The indirect tests were administered from this tape but the tester mimicked his actions for the direct test so that he furnished no clues to the subject.

The masker level for all tests was 75 dB SPL as read on a B & K type 2203 sound level meter using a 6cc artificial ear with headphone

TABLE 6

Organization of Pilot Test

A. Presentation Specifications

Item	Noise	Signal	Both
Duration (msec)	400	200	400
Trans. Shape	Expo.	Expo.	.....
Rise-Fall (msec)	40	40	.....
Delay (msec)	0	100	.....
Sig/Noise (dB)	.....	.....	-4 to 4
Frequency (Hz)	800	800	800

B. Trial and Run Specifications

Item	Duration (sec)	Number
Trial-Overall	6.4	.....
Inter-present.	0.6	.....
Inter-trial	5.0	.....
Pres./Trial	.....	2
Trials/Run	.....	100
Runs/Subject	.....	6
No. of Subjects	.....	4

tension set at 5 ounces. The tests were taken monaurally using TDH-49 earphones with MX-41/AR cushions. The subject indicated his choice of interval by entering either a "1" or "2" in the appropriate box on a form. No information as to score was given to the subject during the tests. Six tests were administered to each of four subjects.

Table 7 displays the results of a 2-way analysis of variance for ch of the four subjects. In this analysis, the row results reflect the signal to noise ratios across which each subject was tested, the column results reflect the taped condition, ie. signals taken from the tape recorder, as compared to the direct condition, ie. signals delivered to the subject directly from the signal generators. Each run contained 5 signal to noise ratios ranging from +4 to -4 dB, with each ratio repeated 20 times at random. Each run was replicated 3 times so that each cell of the analysis contained three replications. The denominator of the F-ratio calculations was the within-cell variance. The results of the analysis do not permit the rejection of the null hypothesis which holds that there is no significant difference arising from the use of either taped or direct signals.

TABLE 7

F-Ratios: Pure Tones--Direct vs Taped

	F-Ratio			D.F.			
Sub	Column	Row	Int'n	Row	Col	Int	Cell
1	.200	90.05	.050	2	1	2	12
2	.600	28.83	.112	2	1	2	12
3	.372	27.42	.094	2	1	2	12
4	1.75	30.27	.502	2	1	2	12

Headings

Column.....Taped vs Direct modes

Row.....Signal to Noise Ratios

Int'n.....Interaction

*Note "S_w" the within cell variance was used as the denominator in finding all F-ratios.

CHAPTER IV.

MAIN INVESTIGATION: THE DETECTION OF TONES IN VARIABLE BANDWIDTH NOISE

I. Procedures

The main investigation, also, uses a two alternate forced choice paradigm (2ATFC) with a temporal organization similar to the pilot test except for a slightly shorter quiet period between trials and ramp transitions instead of exponentials. This organization is given in Table 8, along with other specifications.

Each run comprises 160 trials, the first 20 of which are practice trials and are not scored. Within each run, a range of signal to noise intensity levels from -15 to +11 dB was covered in steps of 2 dB with each ratio repeated 10 times and randomly distributed throughout the run. Each run was taped only once but, in testing, it is replicated 4 times per subject giving a total of 40 replications per data point per subject per noise type per bandwidth. There are 2 taped runs per noise bandwidth, each using a different frozen noise, and 10 bandwidths progressing in octaves from 1.25 to 640 Hz. The total number of trials is therefore, 12,800 trials per subject of which 11,200 are scored. For all 5 subjects, this comprises 56,000 trials.

The frequency of the signal and the center of the noise bands is 800 Hz as noted in Table 8 and the SPL of the noise is always 75 dB measured under the same physical conditions as in the pilot test.

TABLE 8

Organization of Main Investigation

A. Presentation Specifications

Item	Noise	Signal	Both
Duration (ms)	400	200	.....
Trans. Shape	Ramp	Ramp	.....
Rise-Fall (ms)	40	40	.....
Delay (ms)	0	100	.....
Mid-Frequency	800	800	.....
Bandwidth (Hz)	1.25-640	.....	.....
Band Shape	4th order Butterworth	.....	.....
Sig/Noise (dB)	.....	.....	-15 to 11
Onset Phase (degrees) Sig re: Noise	0	90	.....

B. Trial and Run Specifications

Item	Duration (sec)	Number
Trial-Overall	5.4	.....
Inter-present.	0.6	.....
Inter-trial	4.0	.....
Pres./Trial	.....	2
Trials/Run	.....	160
Runs/Subject	.....	80
No. of Subjects	.....	5

The physical testing set-up, sound room, earphones, cushion, etc. are also identical to those used in the pilot test.

Five subjects with normal hearing, in the test ear were used of which three were female, and two were male. Four of the subjects could be described as experienced listeners; the remaining one, a female subject, was clearly naive, never having previously participated in a psychoacoustic experiment. Subjects 1 and 2 were near age 50, the others, near 30 or below.

Since the listening time per run was 15 minutes, taking 4 replications contiguously required a full hour of listening and was tiring. Except for the naive listener, who was also the only paid subject, the tests extended over approximately 6 weeks (some subjects taking less or more time) in which period, subjects devoted 1 to 3 hours per day, 3 to 5 days per week. The paid subject devoted 4 hours per day, 3 days per week until completion. Total listening time per subject was 20 hours.

The subjects indicated their choice of interval, as they did for the pilot test. The ideal arrangement, entry of the responses directly into the computer for immediate processing, was not feasible. Instead, the subject's choices were scored by hand immediately after each run as a check on entry errors. Subjects were also interviewed after each set of 4 runs regarding their subject reactions.

II. Signal Generation

A. General

The signals, pure tones and noises, used in the main investigation were generated by computer and peripherals using the programs

described in Chapter I and stored on audio tape. The ensemble of 20 masking noises was computed only once and stored on digital tape for subsequent recall, modification and combination with the signal, using the double pass technique also described in Chapter I.

B. Specification of Parameters

The specifications for both the pure tone signals and the masking noises are given in Table 8. There are two kinds of specifications; Those which are read in at the beginning of each run (fixed), and those which vary with each presentation (variable).

The fixed specifications comprise, for noise, the probability density distribution, center frequency, bandwidth, type of filter, supression interval, duration, rise and fall shapes, rise and fall times, and level. For signals, the fixed specifications include frequency, duration, rise and fall shapes, rise and fall times, and the onset delay relative to the initiation of the noise. The variable specifications per run are only the signal to noise ratio in dB, and the designation of the presentation within the trial in which the signal is to occur. The rationale underlying the choice of specifications is discussed below.

Center Frequency

Eight hundred Hz was chosen for the signal and the center of the noise band, solely because the investigations most closely related to this study were made at or included this frequency (Schafer et al, 1950; Hamilton, 1957; den Brink, 1963). Also, other related investigations were made at the nearby frequency of 1000 Hz (Fletcher, 1940;

Bos and de Boer, 1966).

Probability Density Distributions

The Gaussian distribution was chosen for similar reasons. Most previous investigations used noise generated by hardware sources and subsequently filtered. Such noise is Gaussian when generated and remains Gaussian after filtering (for instantaneous values not for envelope amplitudes). This last characteristic, the invariance of the Gaussian distribution with linear processing is a useful characteristic and may not be true for other distributions. In addition the Gaussian distribution appears to be a good description of many naturally occuring noises, thermal noise, for example.

Range of Bandwidths

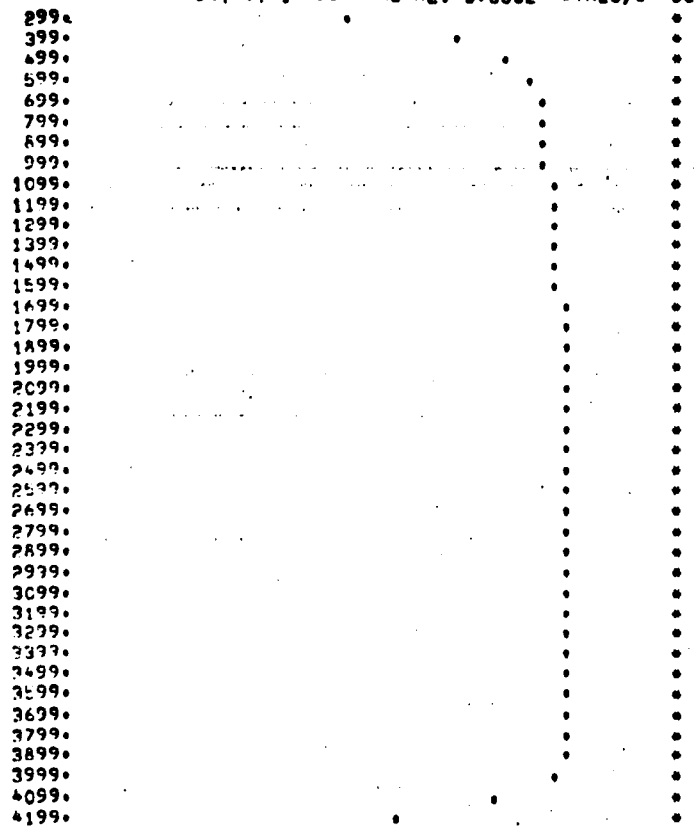
The noise bandwidth range, 1.25 to 640 Hz was chosen as previously noted, to attempt to span the continuum from pure tone detection in wide band noise to pure tone discrimination. Support for the contention that noise with a 1.25 Hz bandwidth may be very close to a pure tone, arises from the experience of the subjects of this experiment who could not distinguish this noise from a pure tone, and from an examination of the envelope of these noises (Figures 23 and 25).

However, though the expectation existed prior to the experiment that the 1.25 Hz bandwidth noise would be indistinguishable from a pure tone, a significant difference becomes evident when a signal is

FIGURE 23: Envelope of Noise A: 1.25 Hz Bandwidth Without Signal

Intensity in dB for Noise A averaged over 100 computer cycles (10 msec.) versus time (10 msec. increments). The left column is the abscissa--increments of time, in msec., and the right column gives the RMS intensity. The display shows the envelope of the noise pulse including rise and fall transitions. (Note Numbering of abscissa starts at 299 for this display.)

OUTPUT IN DB RMS RE. 0.0002 DYNES/CM SQUARED 1.25A

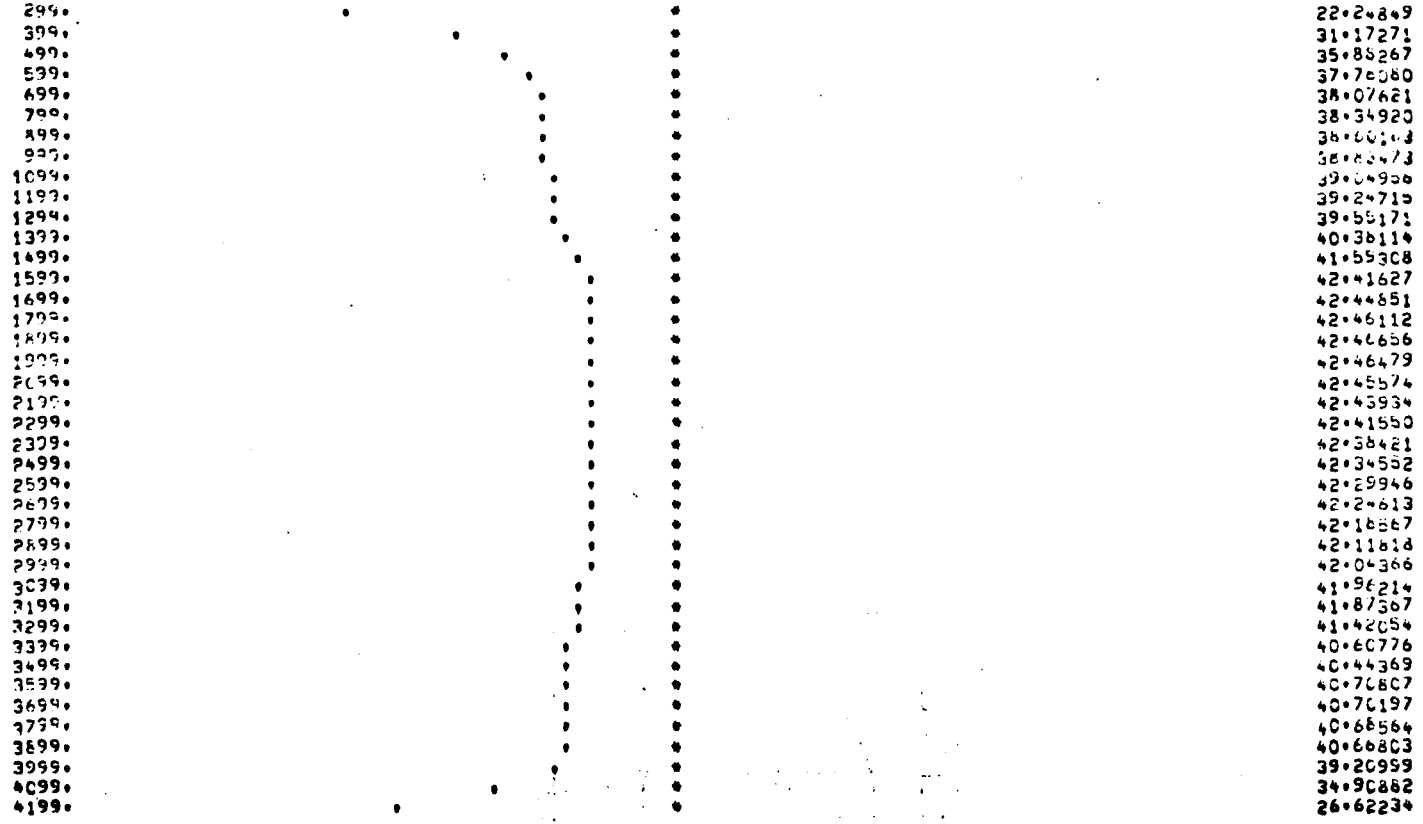


299.	22.24849
399.	31.17271
499.	35.85287
599.	37.76060
699.	38.07621
799.	38.34920
899.	38.60163
999.	38.83473
1099.	39.04956
1199.	39.24715
1299.	39.41857
1399.	39.55483
1499.	39.74686
1599.	39.88541
1699.	40.01113
1799.	40.12460
1899.	40.22642
1999.	40.31722
2099.	40.39760
2199.	40.46813
2299.	40.52929
2399.	40.58164
2499.	40.62571
2599.	40.66203
2699.	40.69120
2799.	40.71374
2899.	40.73022
2999.	40.74112
3099.	40.74696
3199.	40.74822
3299.	40.74530
3399.	40.73871
3499.	40.72898
3599.	40.71658
3699.	40.70197
3799.	40.68564
3899.	40.66803
3999.	39.20999
4099.	34.90482
4199.	26.62234

FIGURE 24: Envelope Of Noise A: 1.25 Hz Bandwidth--S/N = -1 dB

Same as Fig. 23 with the addition of a signal (S/N ratio = -1 dB) delayed 100 msec. in respect to the noise (tone starts at abscissa value of 1299, ends at 3299.) Note numbering of abscissa starts at 299 for this display.)

OUTPUT IN DB RMS RE. 0.0002 DYNES/CM SQUARED 1.25A. S/N -1DB



299.	22.2849
399.	31.1727
499.	35.8826
599.	37.7e080
699.	38.07621
799.	38.34920
899.	38.60[illegible]
999.	38.8[illegible]
1099.	39.04956
1199.	39.2471[illegible]
1299.	39.56171
1399.	40.36114
1499.	41.55308
1599.	42.41627
1699.	42.44851
1799.	42.46112
1899.	42.46656
1999.	42.46479
2099.	42.45574
2199.	42.43934
2299.	42.41550
2399.	42.38421
2499.	42.34552
2599.	42.29946
2699.	42.24613
2799.	42.18567
2899.	42.11818
2999.	42.04366
3099.	41.96214
3199.	41.87367
3299.	41.42054
3399.	40.66776
3499.	40.44369
3599.	40.76807
3699.	40.76197
3799.	40.66564
3899.	40.66803
3999.	39.20959
4099.	34.90882
4199.	26.62234

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FIGURE 25: Envelope of Noise B: 1.25 Hz Bandwidth Without Signal

Same coordinates as Fig. 23. (Note numbering of abscissa starts at 299 for this display.)

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OUTPUT IN DB RMS RE. 0.0002 DYNES/CM SQUARED 1.25 B

299.	*	25.14169
399.	*	33.76857
499.	*	38.14492
599.	*	39.74237
699.	*	39.73155
799.	*	39.72340
899.	*	39.71855
999.	*	39.71731
1099.	*	39.71978
1199.	*	39.72599
1299.	*	39.73591
1399.	*	39.74942
1499.	*	39.76636
1599.	*	39.78648
1699.	*	39.80951
1799.	*	39.83515
1899.	*	39.86307
1999.	*	39.89292
2099.	*	39.92435
2199.	*	39.95705
2299.	*	39.99054
2399.	*	40.02431
2499.	*	40.05785
2599.	*	40.09073
2699.	*	40.12259
2799.	*	40.15293
2899.	*	40.18157
2999.	*	40.20814
3099.	*	40.23235
3199.	*	40.25386
3299.	*	40.27231
3399.	*	40.28733
3499.	*	40.29857
3599.	*	40.30581
3699.	*	40.30885
3799.	*	40.30745
3899.	*	40.30138
3999.	*	38.80630
4099.	*	34.47922
4199.	*	26.09355

TRIAL NUMBER • 4C HAS BEEN TAPED
•EXIT•

FIGURE 26: Envelope of Noise B: 1.25 Hz Bandwidth--S/N = -11 dB

Same as Fig. 25 with the addition of a signal (S/N ratio = -11 dB). (Tone starts at abscissa value of 1299, ends at 3299. Note numbering of abscissa starts at 299 for this display.)

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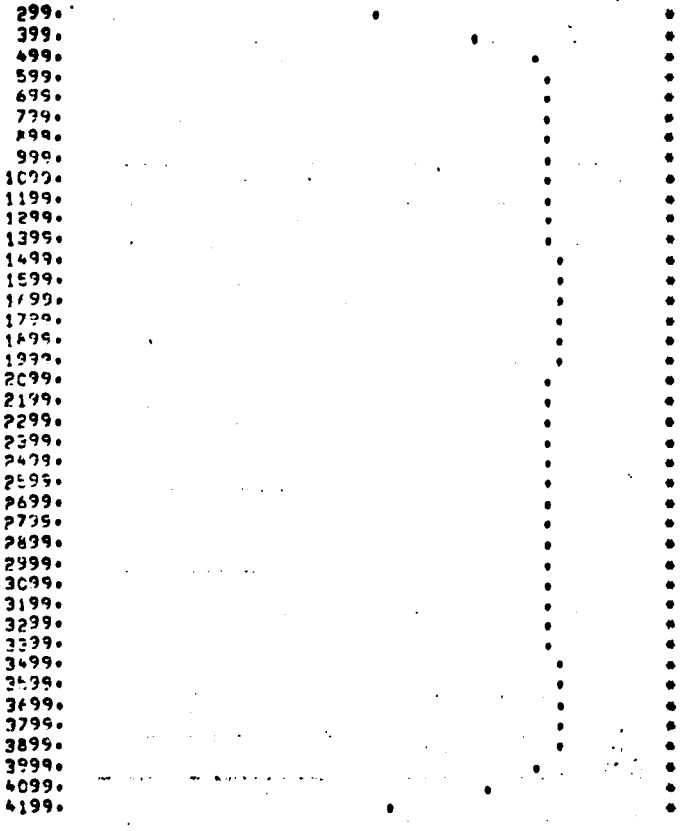
OUTPUT IN DB RMS RE. 0.0002 DYNES/CM SQUARED 1.25B S/N -11 DB



299.	25.14169
399.	33.76857
499.	38.14492
599.	39.74237
699.	39.73155
799.	39.7[illegible]
899.	39.71[illegible]
999.	39.71731
1099.	39.71978
1199.	39.72599
1299.	39.76419
1399.	39.87712
1499.	40.03007
1599.	40.11570
1699.	40.08348
1799.	40.05336
1899.	40.02585
1999.	40.00068
2099.	39.97763
2199.	39.95645
2299.	39.93679
2399.	39.91819
2499.	39.90025
2599.	39.88258
2699.	39.86477
2799.	39.84651
2899.	39.82749
2999.	39.80745
3099.	39.78622
3199.	39.76350
3299.	39.74[illegible]
3399.	39.92603
3499.	40.15587
3599.	40.30597
3699.	40.30685
3799.	40.30745
3899.	40.30138
3999.	38.80630
4099.	34.47922
4199.	26.09355

added. This difference arises from the fact that the phase of the noise, though changing slowly, may vary enough during signal presence to shift the phase relationship from quadrature to opposition. Though this does not occur for Noise A (Figure 24) it does occur for Noise B (Figure 26) wherein the initial increment due to the addition of the signal shifts gradually to decrement (refer to the ordinate values rather than the plotted points which do not have sufficient resolution to show a gradual change).

Selectivity

Typically, filters are chosen to provide an acceptable compromise among the factors selectivity, rise time, and overshoot ripple. For a fixed 3 dB bandwidth, all these factors increase as the order of the filter increases. However, for the computer generation, program only selectivity is the governing requirement, since filter transients are permitted to subside before the filter output is used. A quadruple-tuned Butterworth filter was chosen for this study on the basis that it gave acceptable selectivity and settling time. (time required for transients to subside.) The characteristics of this filter, along with filters of other orders, are shown in Table 9.

Duration and Delay

The choice of the duration of the noise presentation resulted from a two-way compromise between the desire to investigate sub-critical detection at near optimum sensitivity and a need for limiting the duration of each presentation to reduce memory effects during a

trial. A duration of 400 msec. appeared to be a good compromise with a 100 msec. onset delay and a termination 100 msec. prior to that of the noise.

A number of investigators have shown that signal duration has small effect upon detection sensitivity for durations exceeding 150 to 200 msec. (Miller, 1948; Green, Birdsall, and Tanner, 1957; Miskolcsy-Folder, 1959; Green, McKey, and Licklider, 1959; Plomp and Bouman, 1959; Zwislocki, 1960). Though some of these investigations were conducted in quiet those of Green, Birdsall, and Tanner and Green, McKey and Licklider, measured temporal summation for tones in wide band noise.

Onset delays between pulsed noise and tones were found not to affect the course of temporal summation by Zwicker and Wright (1963) for temporal test conditions encompassing those of this main investigation. However, Hamilton (1957) using delayed tones masked by pulsed noise reported improvements in sensitivity with signal duration to 400 msec. but den Brink (1964) who used continuous masking noise found only small improvements beyond 200 mseconds.

The findings, reported above, argue for a signal duration of at least 200 msec or possibly longer to 400 msec. However each increase in signal duration requires similar increases for the noise because of the need to retain a sufficient onset delay to prevent transient masking of the tone (overshoot). Though the literature on overshoot is reassuring, indicating that it diminishes as signal duration increases and does not occur, to begin with, for sub-critical bandwidths (Zwicker, 1965), it was considered advisable, nevertheless, to further

minimize this possibility by using onset delays of 100 msec. for 3 reasons.

The first reason is supplied by Zwicker (1964) who found overshoot effects of 2 to 3 dB for a tone 200 msec. long delayed 2 msec. relative to a wide band masker. He reported also that this effect disappears for noise bands one critical band wide. Since this main investigation includes both supra- and sub-critical noise, it was considered advisable to provide a 100 msec. onset delay to reduce this small effect.

Though Zwicker reported that overshoot disappeared for masking noises equal to, and presumably less than, one critical band wide, it should be noted that, in his work, he used a 2 msec. long, 5 KHz tone burst passed through critical and supra-critical bandwidth filters. Note that a 2 msec burst has a main spectrum lobe of 500 Hz which is approximately equal to the 5 KHz critical bandwidth. On this basis, his conclusion that the degree of overshoot depended on the dissimilarity of signal and noise spectrums is supported by his finding that overshoot disappeared at noise bandwidths of 1 critical band. If, however, this hypothesis holds, then overshoot could persist to sub-critical bandwidths for the present investigation since a 200 msec. pulse had a main lobe dispersion of only 5 Hz. This was the second reason for introducing a 100 msec. onset delay.

A third reason related to the second but not fully appreciated in the planning stage arose from the realization that under some conditions subjects were listening for changes in the noise as clues to

signal presence. This required a period of noise preceding the region in which the signal might appear.

Because of these considerations, the temporal structure of each presentation resolved into a noise sample 400 msec long, a signal, when present, 200 msec. long added to the noise with a 100 msec. delay. All durations are measured from initiation to termination and delays, from initiation to initiation.

Noise Sample

The reasons underlying the decision to use frozen noises have already been discussed. The requirement that the initial filtering transients had subsided before the noise was used, established a minimum supression period of 16,000 computational cycles (1.6 seconds of audio). The rise time for the narrowest bandwidth, 1.25 Hz, is approximately .6 seconds so that a minimum interval equal to 2.5 rise times occurs before the noise is switched to output. Table 9 shows that by this time the noise has reached steady state.

These considerations dictated the minimum supression time. Since no specific theoretical considerations were applicable to the decision, Noise A was chosen on the basis of the minimum suppression time, and Noise B was taken contiguously.

Transitions

The ramp transition shown schematically in Figure 13-D and empirically in Figure 27 was chosen for both noise and signal. This transition gives good concentration of spectral energy within the primary

TABLE 9

Stagger-Tuned Butterworth Filters

Type of Network	t/s	Delta	B*Tau
Synchronous	n	0.0	0.7
Pair	1	4.3	0.67
Triplets	1	8.15	0.73
Quadruple	1	10.9	0.78
Quintuple	1	12.8	0.82
Sextuple	1	14.3	0.85
Septuple	1	15.4	0.89

"t" = no. of identical stages

"s" = detuned circuits

Delta = % overshoot

"B" = total bandwidth

Tau = rise time in msec.

.....

FIGURE 27: Envelope of Noise Including Transitions

Intensity in dB for Noise B averaged over 100 computer cycles (10 msec.) versus time (10 msec. increments). The left column is the abscissa--increments of time, and the right column gives the RMS intensity. The display shows the envelope of the noise pulse including rise and fall transitions.

OUTPUT IN DB RMS RE. 0.0002 DYNES/CM SQUARED 1.25 B

299.	25.14169
399.	33.76657
499.	38.14492
599.	39.74237
699.	39.73155
799.	39.72340
899.	39.71855
999.	39.71731
1099.	39.71978
1199.	39.72599
1299.	39.73591
1399.	39.74942
1499.	39.76636
1599.	39.78648
1699.	39.80951
1799.	39.83515
1899.	39.86307
1999.	39.89292
2099.	39.92435
2199.	39.95705
2299.	39.99064
2399.	40.02431
2499.	40.05785
2599.	40.09073
2699.	40.12254
2799.	40.15293
2899.	40.18157
2999.	40.20814
3099.	40.23235
3199.	40.25386
3299.	40.27231
3399.	40.28733
3499.	40.29857
3599.	40.30581
3699.	40.30885
3799.	40.30745
3899.	40.30138
3999.	38.80630
4099.	34.47922
4199.	26.09355

TRIAL NUMBER = 40 HAS BEEN TAPED
•EXIT•

CALCOMP

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lobe and permits a clear definition of initiation and termination of the noise and signal pulses and transition regions since it begins and ends with definition in the amplitude-time domain. The overall duration of pulses were measured from the initiation to the termination of each transition.

The Fourier transforms of the envelope of the pure tone signal pulse with ramp transitions were calculated for various transition times. The decision to use a 40 msec. transition time was based on these results which showed that the energy spilled beyond the narrowest bandwidth of the main investigation, decreased slowly beyond this point. Figure 28 displays the spectrum of the modified pulse and Figure 29 gives the proportion of energy spilled beyond the abscissa frequency deviation.

The literature of psychoacoustic testing also indicates that a 40 msec. transition time is sufficient to eliminate clicks but since the ramp transition has not been used a brief listening test was conducted to determine if clicks were audible. The results supported the literature; no clicks were heard.

Level

Questions about the appropriate noise level, or levels at which sub-critical detection should be investigated, devolve to other questions about the stability of the critical bandwidth and the course of sub-critical threshold as a function of noise level. There is wide, but

FIGURE 28: Spectral Dispersion of the Envelope of the 200 msec. Signal Pulse with Rise and Fall Transitions of 40 msec. The left column gives the cycles above center frequency; the right column gives the corresponding amplitude ratio of the spectral density curve compared to the "0" Hz point. The curve is mirrored for cycles below center frequency.

SPECTRAL DISPERSION

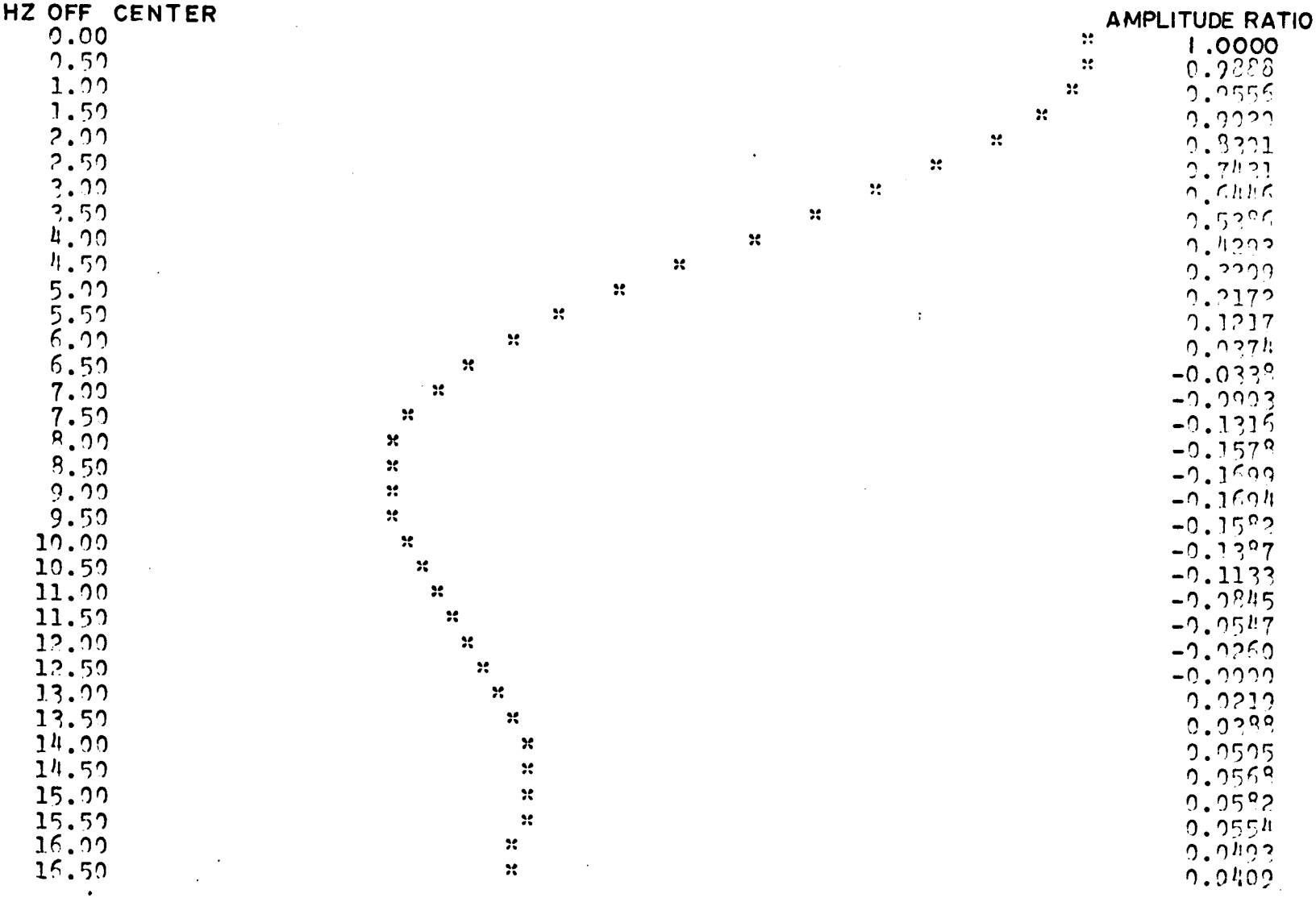
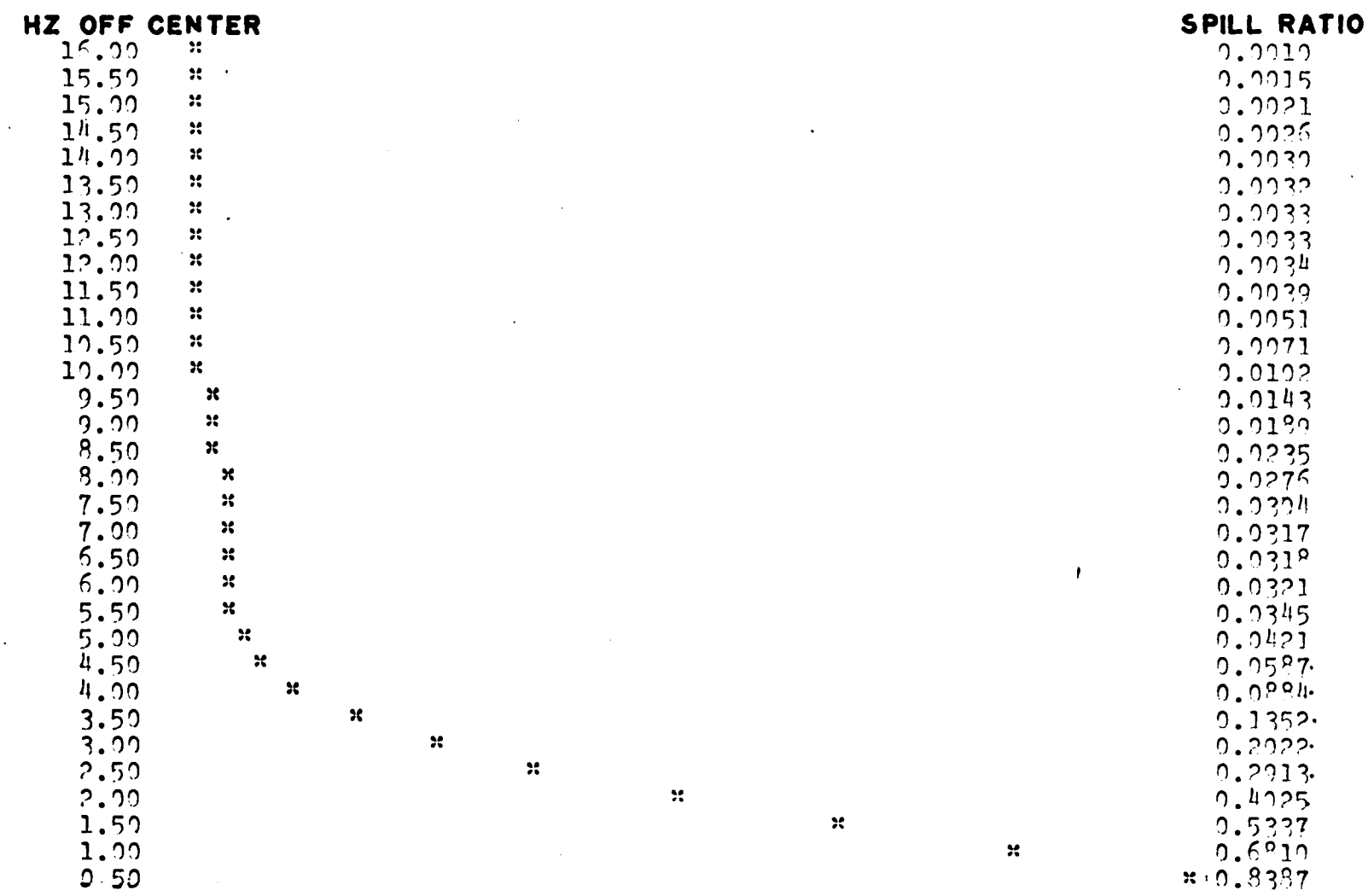


FIGURE 29: Energy Spilled by the 200 msec. Signal Pulse with rise and fall transitions of 40 msec. The left column gives the cycles off center frequency (abscissa); the right column (ordinate) gives the decimal ratio of the total pulse energy which is contained in all frequency components exceeding the abscissa frequency.

ENERGY SPILL



far from complete, agreement that the critical band does not change with level.

Supporters of the concept that it is unaffected by level include Fletcher (1940), Hawkins and Stevens (1950), Greenwood (1961a) and den Brink (1964). However, Bourbon, Evans and Deatherage (1968) and Canahl (1971) present evidence that the critical band widens with increasing masker level.

Den Brink (1964) supplied information regarding the stability of the sub-critical threshold with masker level since his investigation was conducted at three levels of masking noise, 35, 50 and 65 dB SPL. He reported "no significant differences between the measurements in terms of signal to noise ratio," and pooled data taken at all three levels.

In view of den Brink's findings, the preponderance of support for an invariant critical band, the lack of evidence presented by the opposition that the essential trends of subcritical thresholds are affected by level changes, and the logistical limitations on computer time, tapes and testing time, the decision was made to test at one noise level.

The 1000 Hz calibration tone, recorded at the beginning of each run precisely 15 dB above the noise, was set at 90 dB SPL as measured on a B&K, Type 2203, sound level meter using a TDH-49 phone with a MX-41/AR cushion, coupled through a 6cc coupler into a B&K condenser microphone under a pressure of .5 ounces. The noise level is therefore 75 dB SPL.

The range of signal to noise ratios, -15 dB to 11 dB in steps

of 2 dB, was established on the basis of a pilot test which followed the procedures of the main investigation closely but used a greater range, -15 to 15 dB. Since ratios above 11 dB were almost always detected perfectly, they were eliminated, permitting an increase in the recorded level of noise and an accompanying increase in the signal to extraneous background noise of the tape recorder. The output of the D-A converter was recorded on a Tandberg tape recorder, Model 9 and played back on an Ampex 4020256. Both tape recorders have wow and flutter ratings below .13% RMS and signal to noise ratios, 55 dB or better.

III. Analysis of Data

A. Preliminary Analysis

Raw Data

The raw data collected in the main investigation consisted of 160 responses per run in the form of either a "1" or a "2" representing a vote for either the 1st or 2nd presentation of each trial as the one containing the pure tone signal. A sample form with responses is shown in Figure 30. As each run was replicated 4 times per subject, each subject produced 4 such forms per run for 20 runs, totalling 12,800 responses for each of 5 subjects. Since subjects, during the pilot testing had expressed a need for time to adjust to the listening situation, the first 20 responses of each run were not scored. As noted earlier, the results of each run were scored immediately following the testing and reduced to a set of proportions for each of the 14 signal to noise ratios. The 4 replications of a given noise sample-bandwidth

FIGURE 30: Sample Form Used to Collect Data

for both pilot test and main investigation. Subjects indicate choice of either the first or second presentation within a trial, by writing either a "1" or a "2" within each box. Numbered section on the right is used to summarize the proportion correct at appropriate signal to noise ratios.

DETECTION OF SIGNALS IN NOISE
M. ROSENSTEIN

SUBJECT RESPONSES

NAME..M.R..... TRIALS 1-320.
FREQUENCY 1-10. 11-B

1	2	3	4	5	6	7	8	9	10
1	2	2	1	1	2	1	2	2	1
1	1	2	2	1	1	1	2	2	1
1	1	2	1	1	2	2	2	1	1
1	2	2	2	2	2	1	1	1	1
1	1	2	1	1	2	1	2	2	1
2	1	1	2	1	1	2	2	1	2
2	2	2	1	1	1	2	1	1	1
1	2	2	1	1	2	2	1	2	1
1	1	2	2	1	2	1	2	2	1
2	2	2	1	1	1	2	1	2	1
1	2	2	2	1	1	1	2	1	1
2	2	2	1	1	2	1	2	1	1
2	1	2	1	2	2	1	1	1	1
1	2	2	2	2	1	1	2	1	1
2	2	2	1	1	1	2	1	2	1
1	1	1	2	1	1	2	1	1	1

11.....
9.....
7.....
5.....
3.....
1.....
-1.....
-3.....
-5.....
-7.....
-9.....
-11.....
-13.....
-15.....

combination produced 4 such sets based on 10 responses per proportion for each subject for each noise-bandwidth. These data were entered into digital data files and provided the data for statistical analysis.

Pooling Data

The first analysis investigated the validity of pooling the 4 replications of each noise-sample-bandwidth combination for individual subjects. A two way analysis of variance with 14 rows (signal to noise ratios), 4 columns (replications), and 1 entry per cell was used. On the first pass, the null hypothesis that no significant differences existed between columns was sustained for all but a tiny percentage of sets. Inspection of these deviant sets indicated that in most cases habituation or learning had affected the first run. Deletion of this run resulted in support of the null hypothesis. In the two of the three remaining cases some factor, possibly drowsiness or inattention, had caused an unusual deviation in data and required the elimination of a run other than the first, to sustain the null hypotheses. In the last remaining case, the subject had reported that after two replications, he had suddenly perceived the pattern of the noise. In view of these results, and after consideration of the testing situation, it was decided to pool the replications on an individual subject basis, after removing the few aberrant replications discussed above.

A second statistical analysis considered the validity of pooling the data from runs with the same bandwidths but with two different noise samples on an individual subject basis. A two way analysis of variance with 14 rows (signal to noise ratios), 2 colums (Noise A and

Noise B), and 4 entries per cell (replications) was made.

In general, the results did not sustain the null hypothesis of no significant difference between the data from Noise A compared with Noise B. Though there was some support for pooling at the narrower and wider bandwidths, the analysis did not support pooling in the intervening region particularly for the 5 to 40 Hz range. In view of these results a decision not to pool was made.

Since Schafer et al, Hamilton, and den Brink pooled their data across subjects and presented averaged results, a third statistical analysis was made to see if this were appropriate for the main investigation. The two way analysis of variance made for each noise-bandwidth combination used 14 rows (signal to noise ratios), 5 columns (subjects) and 4 entries per cell (replications of noise-sample-bandwidth combinations).

Since none of the column "F" ratios supported the null hypotheses of no significant difference between subjects, results were not pooled across subjects. Although in a later section of this dissertation, a rough average curve representing the data of all subjects is used for purposes of comparison with other investigators, no suggestion is intended there that these results are poolable.

Probit Analysis of Data

The determination of the smoothed psychometric curves and the appropriate point on these curves to designate as threshold, was made using probit analysis (Finney, 1971; Natrella, 1963). Probit analysis procedures assume that the data in question, can be fitted by the

normal probability curve (the integral of the normal probability density distrinution). The computer program used here was a modification of an existing program. In addition to obtaining maximum likelihood estimates of the cumulative normal psychometric curves and eveluating the goodness of fit with a Chi squared test, it also plots these curves, selects the threshold values and plots these results versus noise bandwidth as well. A typical psychometric curve showing both the original proportions, and the probits are shown in Figure 31. Table 10 lists the complete results, mean (threshold value), standard error of the mean, variance of the slope, sigma (standard deviation), variance of the mean, variance of the slope, and chi squares resulting from a goodness of fit test between the probit curves and the data, for a typical case. Tables 11 and 12 give the condensed results, mean and sigma only, for all cases. The chi squares resulting from the goodness of fit tests, provide a measure of the validity of the assumption that the psychometric curves can be fitted by the normal probability curve, a basic assumption of probit analysis. Since all but 7 of the 100 psychometric curves, with a maximum of 2 for any one subject, gave chi squares less than that required at the .05 level of significance, this assumption is appropriate.

B. Thresholds vs Noise Bandwidth: Fitted Curves

The thresholds found by probit analysis are shown in Figs. 32-36, in the traditional manner, signal intensity to noise spectrum level ratio plotted against noise bandwidth. Using the technique of orthogonal polynomial analysis, these curves were subjected to further

FIGURE 31: Typical Psychometric Curve Used for Threshold Determination

The abscissa gives the signal to noise intensity ratio; the ordinate the proportion of the correct choices. The "X"s represent the empirical results; the "O"s the best fit probit curve. The threshold is taken as the value of the probit curve at 0.75 on the ordinate. Where "X"s and "O"s coincide, a "+" is shown.

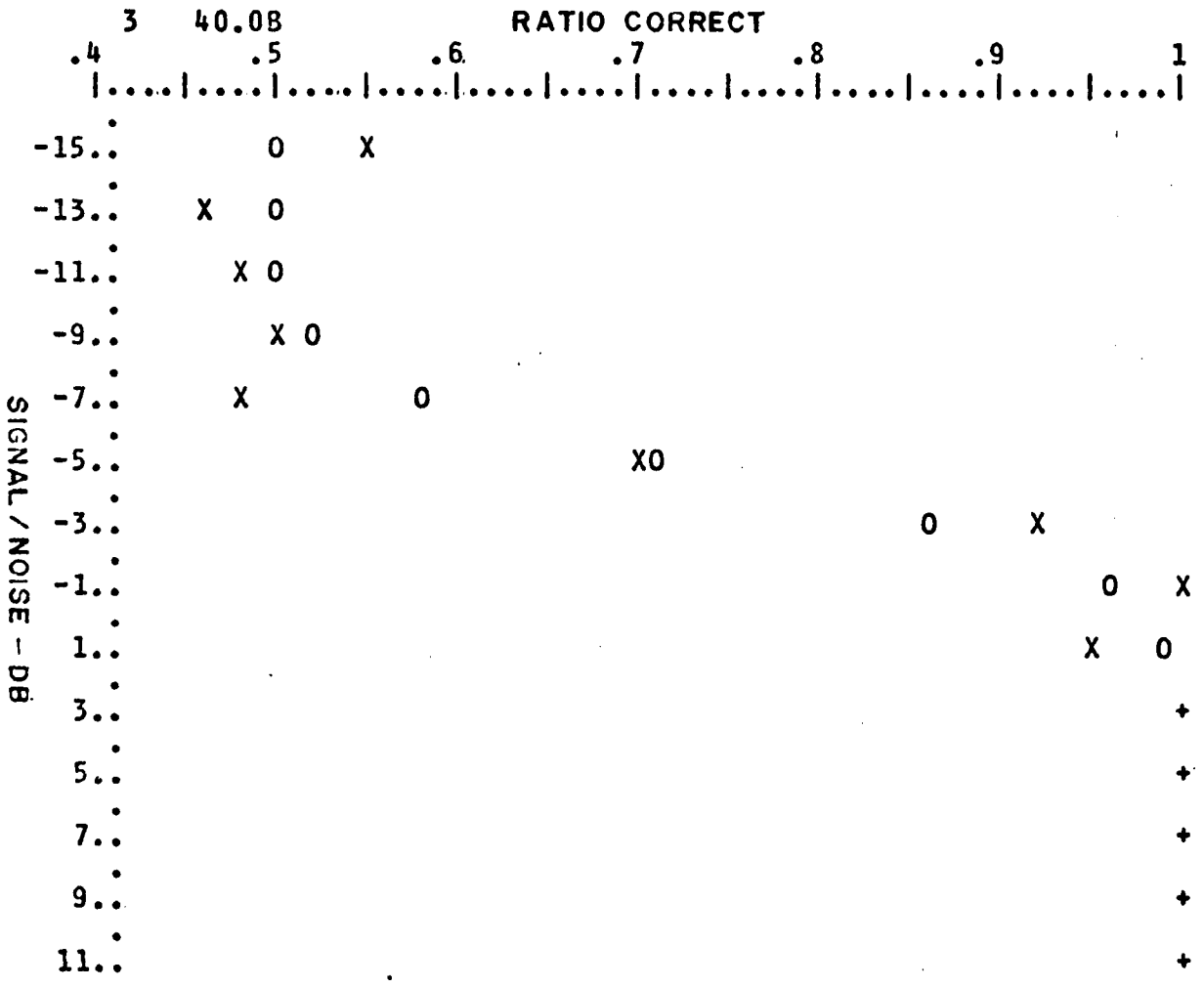

3 40.0B
RATIO CORRECT
.4 .5 .6 .7 .8 .9 1
SIGNAL / NOISE - DB
-15
-13
-11
-9
-7
-5
-3
-1
1
3
5
7
9
11

TABLE 10

Results of Probit Analysis: Complete For Noise 1.25A

Subject	1	2	3	4	5
Band	1.25A	1.25A	1.25A	1.25A	1.25A
Mean	-4.138	-1.978	-3.869	-4.608	-5.014
Stand. Error of Mean	0.475	0.522	0.347	0.662	0.6328
Slope	0.415	0.596	0.692	0.276	0.304
Stand. Error of Slope	0.058	0.177	0.125	0.048	0.056
Sigma	2.408	1.679	1.445	3.628	3.286
Variance: Mean	0.225	0.273	0.121	0.439	0.400
Variance: Slope	0.003	0.032	0.016	0.002	0.003
Chi Squared	19.97	19.143	11.45	16.92	10.37

Chi Squared (.05)=22.36

TABLE 11

Results of Probit Analysis: Condensed --Noise A

Band	Subjects									
	1		2		3		4		5	
	Mean	Sigma	Mean	Sigma	Mean	Sigma	Mean	Sigma	Mean	Sigma
1.25A	-4.14	2.408	-1.98	1.678	-3.87	1.445	-4.61	3.628	-5.01	3.286
2.50A	-6.37	2.896	-3.82	2.052	-11.40	3.58	-7.41	5.037	-7.94	3.576
5.00A	-3.37	3.312	-3.72	1.957	-7.09	3.686	-4.47	3.103	-4.49	2.285
10.0A	-13.98	6.351	+5.93	1.916	+3.37	1.599	-17.83	7.520	+5.05	0.782
20.0A	-0.57	1.292	+3.11	1.702	+2.31	1.904	-17.63	6.952	+2.82	1.231
40.0A	-1.90	2.654	+0.42	2.368	-2.21	3.481	-4.38	2.461	-3.53	4.289
80.0A	-3.90	3.033	-2.02	2.543	-4.76	3.125	-8.78	4.517	-3.19	1.986
160.A	-2.40	4.140	-1.49	2.204	-2.45	3.276	-8.79	5.465	-1.85	3.448
320.A	-2.39	2.265	-3.52	3.220	-4.16	2.529	-4.42	2.320	-1.84	2.212
640.A	-6.98	1.928	-8.21	2.810	-6.57	1.891	-8.65	4.300	-7.05	2.460

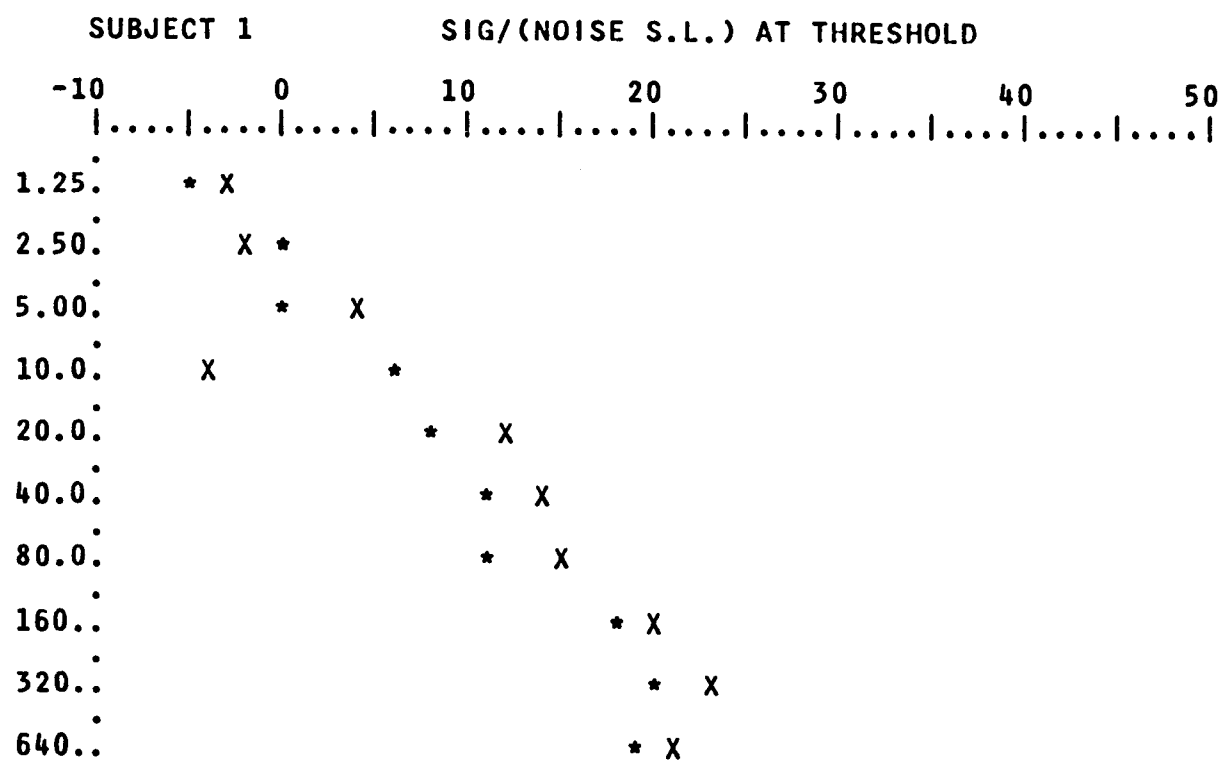
TABLE 12

Results of Probit Analysis: Condensed --Noise B

Band	Subjects									
	1		2		3		4		5	
	Mean	Sigma	Mean	Sigma	Mean	Sigma	Mean	Sigma	Mean	Sigma
1.25B	-5.92	2.273	-1.51	1.418	-7.39	2.718	-8.13	2.593	-7.11	4.660
2.50B	-4.45	4.275	-1.19	2.497	-3.91	1.680	-2.66	2.846	-1.01	2.717
5.00B	-7.32	8.710	+0.57	2.157	-16.62	6.452	-8.78	2.470	-6.30	4.806
10.0B	-4.40	3.765	+2.70	1.452	-7.86	2.913	-7.03	2.904	-3.98	2.744
20.0B	-5.12	4.423	+1.92	4.220	-3.01	1.798	-5.24	3.623	-4.78	2.874
40.0B	-5.08	5.028	-0.74	3.935	-4.49	2.531	-3.11	2.808	-4.16	3.726
80.0B	-7.60	7.354	-1.80	1.707	-2.70	2.611	-12.5	5.907	-6.69	4.325
160.B	-4.39	2.303	-0.87	3.556	-1.33	1.560	-3.66	1.641	-1.60	2.658
320.B	-5.20	2.127	-4.99	3.571	-4.90	2.198	-12.15	5.670	-7.00	4.119
640.B	-8.58	3.568	-6.60	2.683	-8.20	3.882	-10.91	3.890	-5.82	3.637

FIGURES 32 - 36: Threshold versus Noise Bandwidth Curves Computed by Probit Analysis

The abscissa of the figures gives the bandwidth of the noise centered about 800 Hz. The ordinate gives the signal intensity to noise spectrum level at threshold. The "X"s represent the results arising from Noise A; the "*"s, from Noise B.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0
10
20
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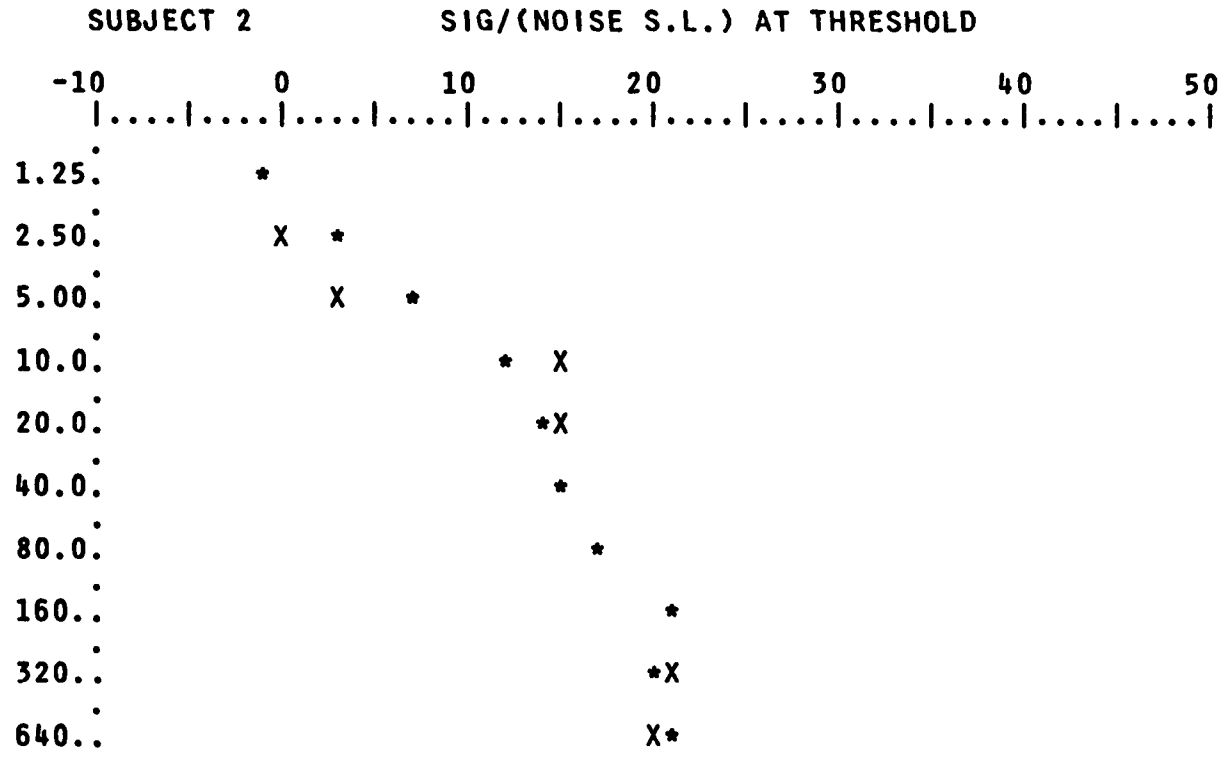
40
50
1.25
2.50
5.00
10.0
20.0
40.0
80.0
160
320
640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SIG/(NOISE S.L.) AT THRESHOLD
-10
0
10
20
30
40
50
1.25
2.50
5.00
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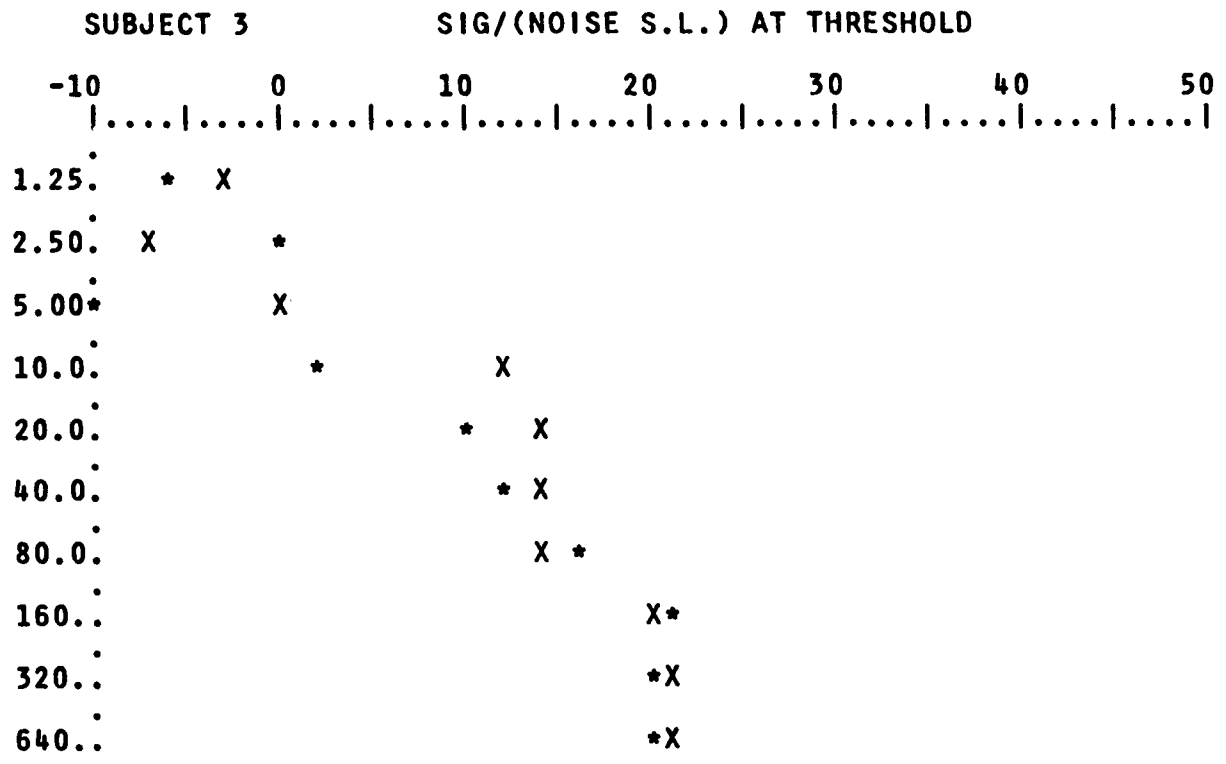
20.0
40.0
80.0
160
320
640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20
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40
50
1.25
2.50
5.00
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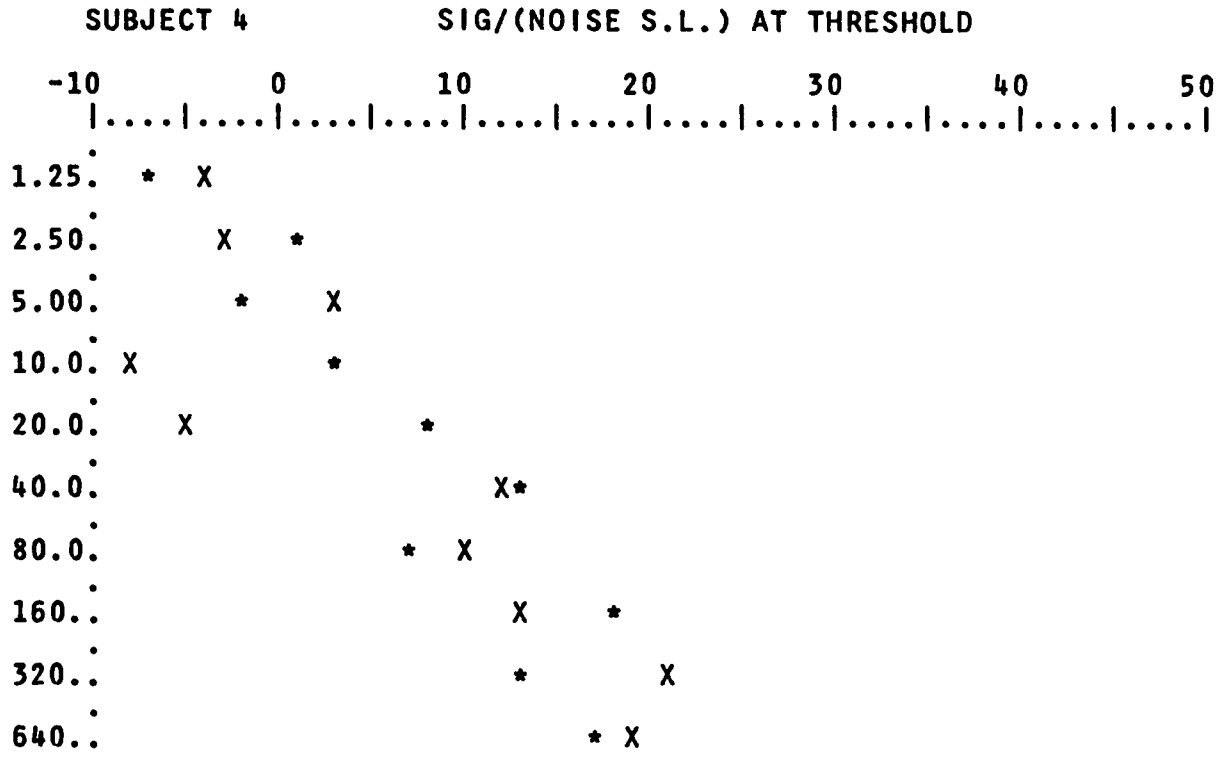
80.0
160
320
640


SUBJECT 4
SIG/(NOISE S.L.) AT THRESHOLD
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0
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40
50
1.25
2.50
5.00
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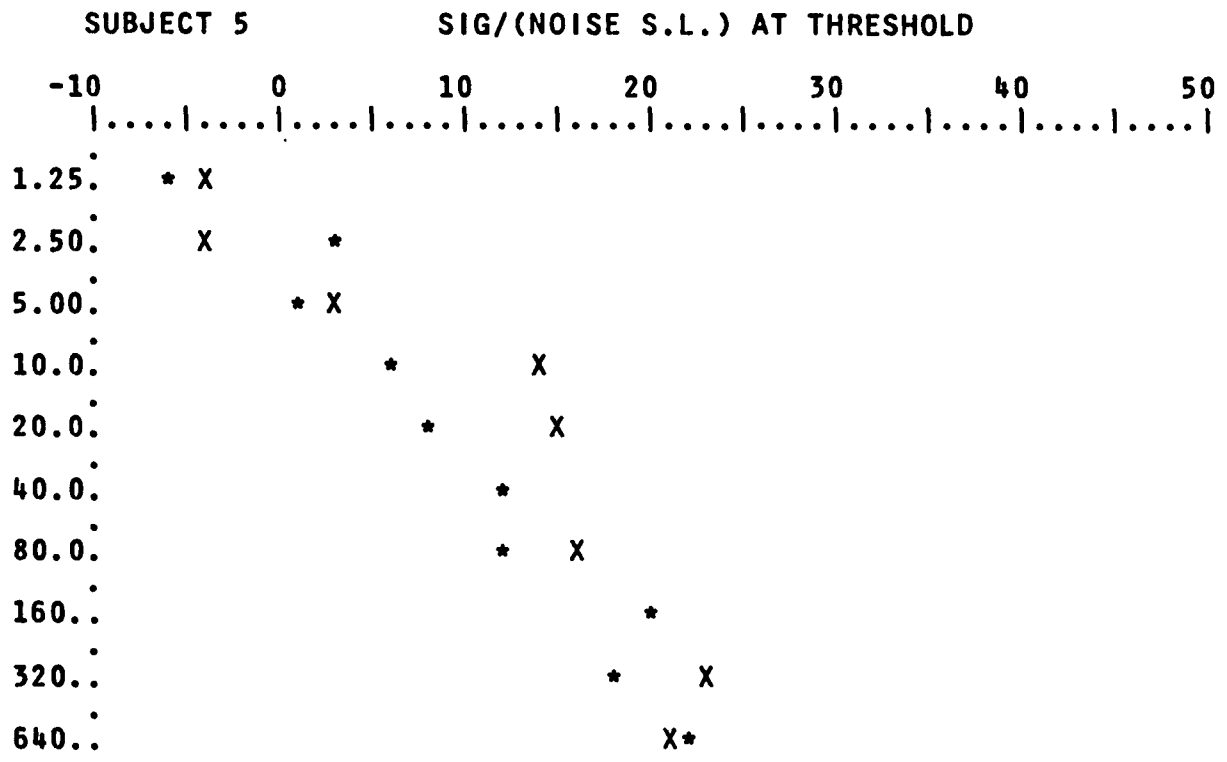
40.0
80.0
160.
320.
640.


SUBJECT 5
SIG/(NOISE S.L.) AT THRESHOLD
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0
10
20
30
40
50
1.25
2.50
5.00
10.0
20.0
40.0
80.0
160
320
640

statistical analysis in order to smooth the results and to expose significant patterns. Since the hypothesis suggested differences in the effects of the two noises due to envelope differences, mainly within the sub-critical noise bandwidth region, the analysis concentrated on these regions. The technique used, is described by Natrella (1963).

To apply this technique, orthogonal polynomials are found term by term. After each term is calculated, an "F" ratio, consisting of the reduction in mean square error due to the last term divided by the overall mean square error, is found. This overall mean square error is the error arising from a comparison of the function consisting of all existing terms of the polynomial and the input data. The order of the best fitting orthogonal polynomial is established by the highest order term giving a significant "F" ratio.

The analysis focussed entirely on the sub-critical region of the function, primarily between 5 and 80 Hz were the divergence of the thresholds produced by the two noises was greatest. Since it appeared that the Noise B sub-critical thresholds could be fitted by a straight line while that of Noise A could not, this departure from linearity was isolated, when it appeared, by restricting the analysis range to the curvilinear region. Since the analysis is based upon signal to noise intensity ratios at threshold rather than upon the signal to noise spectrum level used for the display, a line rising at 3 dB per octave bandwidth in the display is represented by the 0th order term of the polynomial, a straight line rising at a slope other than 3 dB/octave is represented by the 1st order term, and curved lines in the display are represented by higher order terms. Fletcher's hypothesis would be illustrated by a "0th" order polynomial.

The orthogonal polynomial analysis procedure used here, requires equal intervals between noise bandwidth values. There are, however, a few threshold, 1 each for subjects 1 and 3, and 2 for subject 4, which subsequent analysis justifies treating as outlier values. Therefore they were omitted. The two values for subjects 1 and 3 were replaced by a point with a value equal to the average of the two surrounding points. Since the two outlier values for subject 4 were adjacent and in the most critical section of the Noise A curve, the analysis was abondoned in this case.

The results of this analysis are displayed in Tables 13 and 14. The sub-critical section of Noise B thresholds are best fitted by the 0th order polynomial for all subjects except 2. This means that the trend of the result assumed by Fletcher (1940), a straight line rising with a slope of 3 dB per octave noise bandwidth for sub-critical bandwidths, is generated by Noise B for 4 out of 5 subjects (subject 3 deviated slightly over range 1.25 to 160 Hz with a slope of 3.69 dB but gives 3 dB from 5 to 80 Hz). However, Fletcher's equal energy assumption which would require these curves to have ordinate values of 0 dB at 1 Hz, is not supported.

The Noise B thresholds produced by subject 2 are clearly anomalous and cannot be fitted over any sub-critical section by a straight line. It is fitted, however, by a curvilinear polynomial of order "4" over the range 1.25 to 160 Hz. In this case Fletcher is not supported. Figures 37-44 show the threshold functions arising from Noise B together with the best fitting orthogonal polynomials.

No Noise A threshold function can be fitted over any non-trivial sub-

TABLES 13 and 14. Fitted Orthogonal Polynomials

Column 1 lists the subjects by number. Column 2 shows the noise bandwidth range within which the analysis was restricted. Column 3 indicates the order of the highest significant term of the analysis. Column 4 lists the coefficient of each term in ascending order starting with the "0th" term. Column 5 gives the "F" ratio required for a significance level of .05. Note that some curves have been analyzed more than once and over different noise bandwidth ranges.

TABLE 13

Fitted Orthogonal Polynomials
Noise B

Subj.	Range-Hz	Highest Order	Coefficients 0 to N	F-Ratio	F(.05)
1	5-80	0	-5.9037	82	7.71
1	1.25-160	0	-5.534	145	5.59
2	1.25-160	3	+0.1145	.....	.....
			-0.0395	.....	.....
			-0.2517	.....	.....
			+0.0481	29.75	10.13
3	5-80	0	-4.7901	25.13	7.71
3	1.25-160	1	-4.5732	.....	.....
			+0.6855	6.70	5.99
4	1.25-80	0	-6.3875	28.60	5.59
5	5-80	0	-5.1803	87	7.71
5	1.25-160	0	-4.4522	309.7	5.59

TABLE 14

Fitted Orthogonal Polynomials
Noise A

Subj.	Range-Hz	Highest Order	Coefficients 0 to N	F-Ratio	F(.05)
1	5-80	2	-2.3436	.....	.....
			-0.0971	.....	.....
			-0.6805	34.8	18.51
2	5-160	3	+0.3709	.....	.....
			-0.4388	.....	.....
			-0.7882	.....	.....
			+0.7190	15.27	18.51
3	5-80	3	-1.6747	.....	.....
			-0.0920	.....	.....
			-0.2106	.....	.....
			+1.1258	186.46	161
5	5-80	3	-0.6685	.....	.....
			-0.5983	.....	.....
			-1.6086	.....	.....
			+1.5392	244.72	161

FIGURES 37 - 44: Best Fit Threshold versus Noise Bandwidth Curves--

for Noise B computed by orthogonal polynomial analysis. The abscissa gives the noise bandwidth; the ordinate, the signal intensity to noise spectrum level at threshold. The "X"s give the curves computed from empirical data; the "O"s, the best fit orthogonal polynomial.

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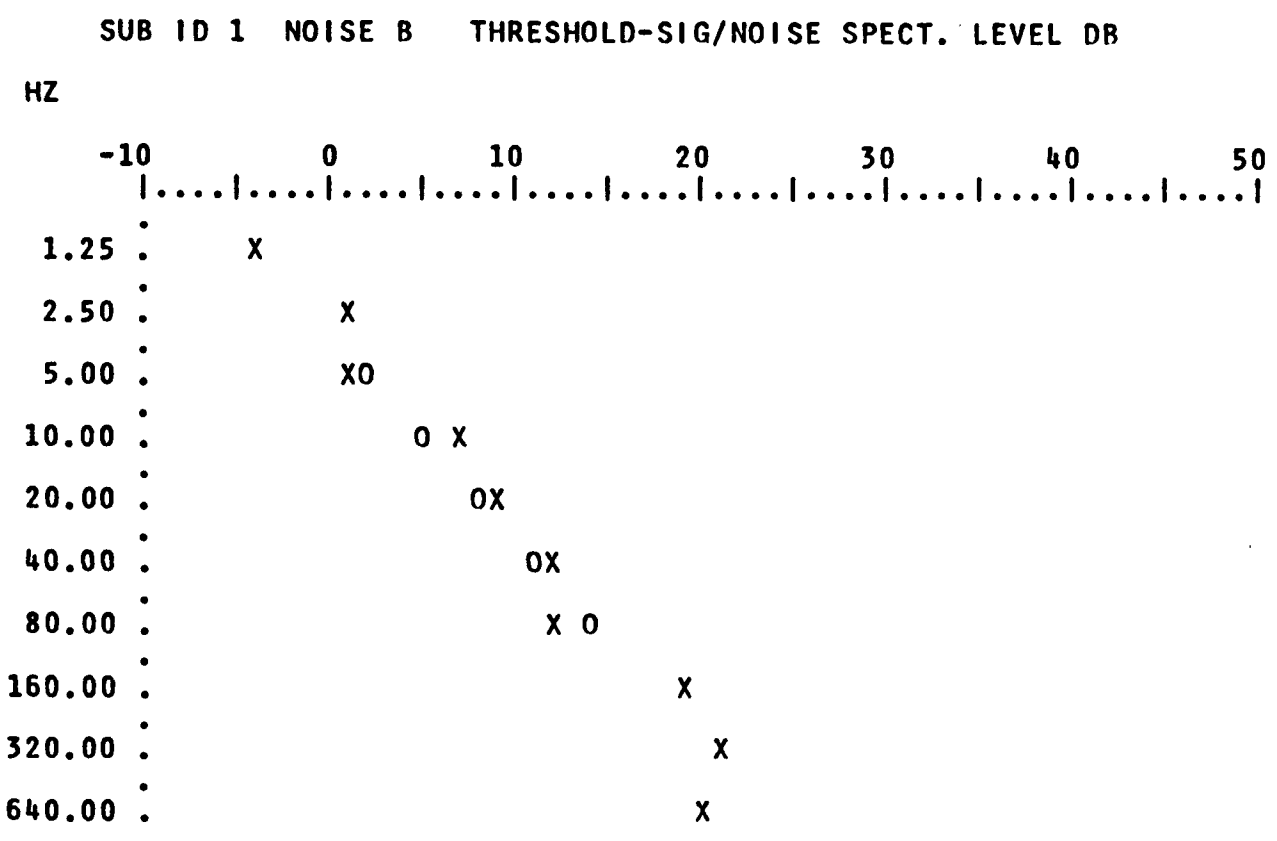

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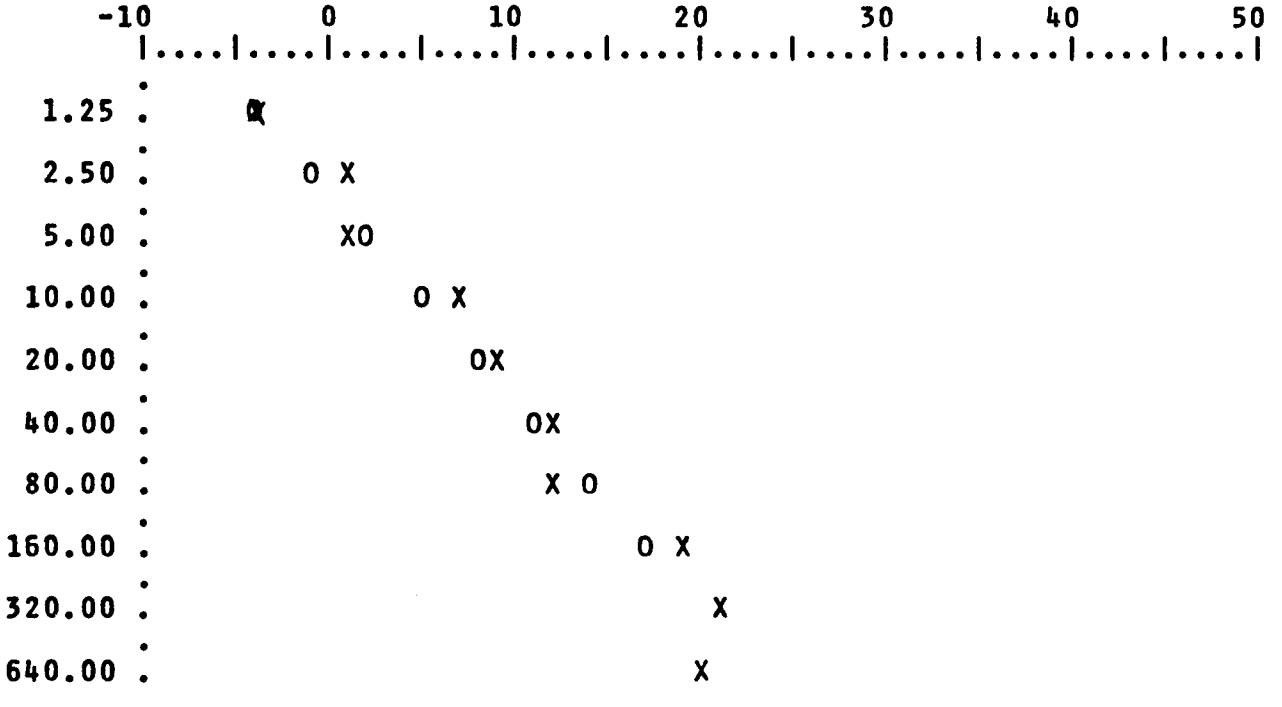

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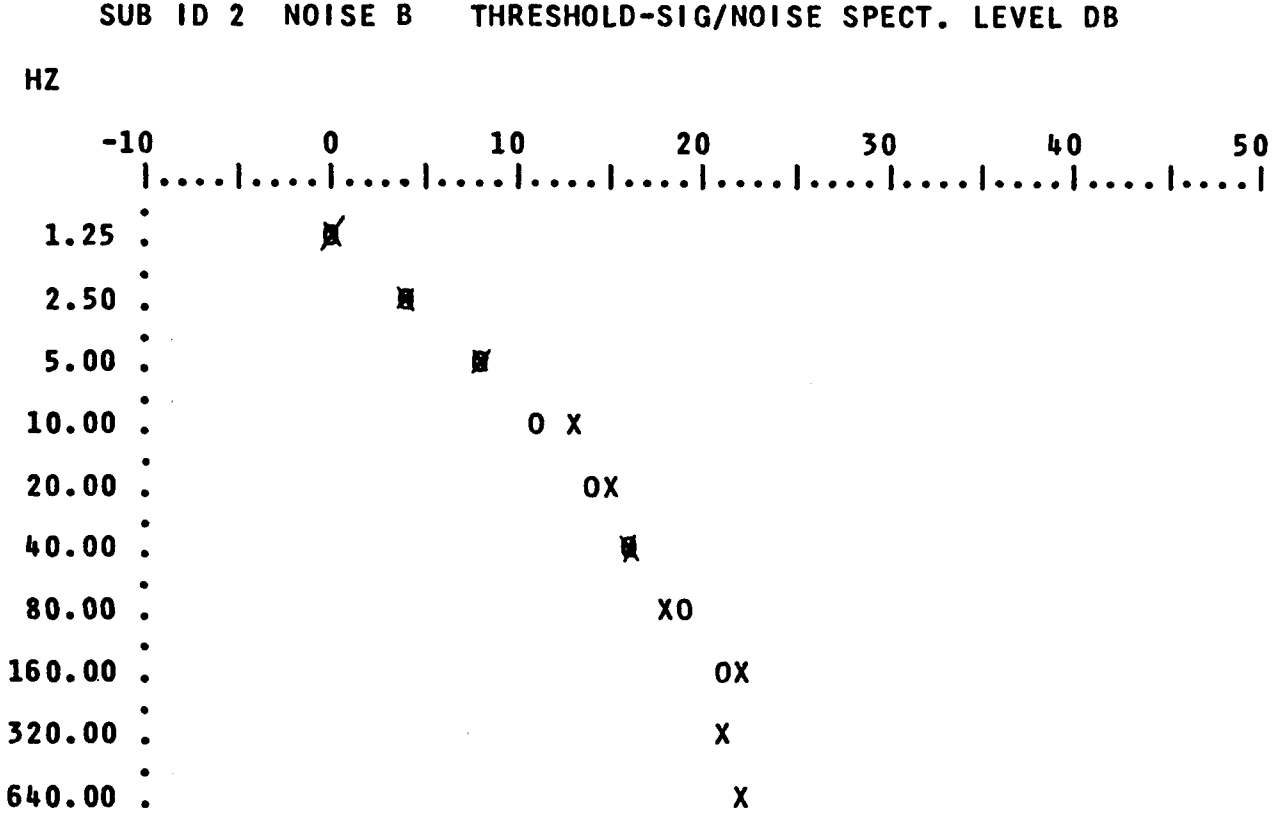

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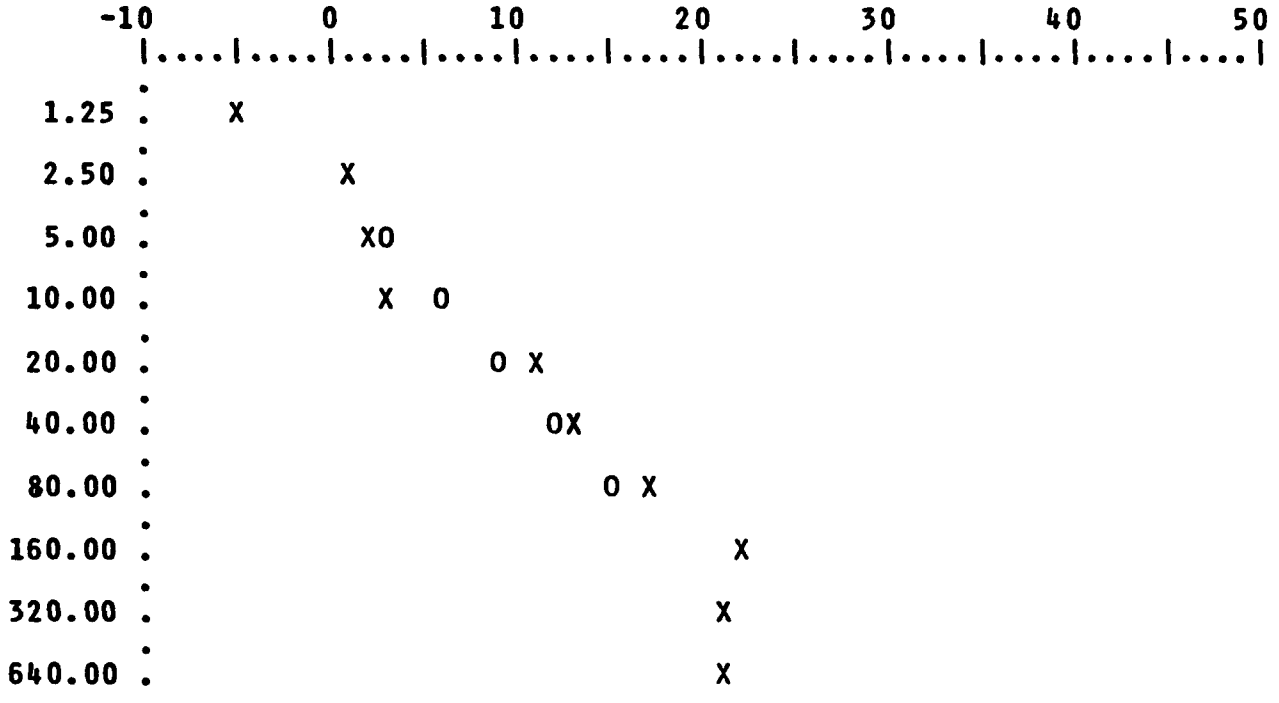

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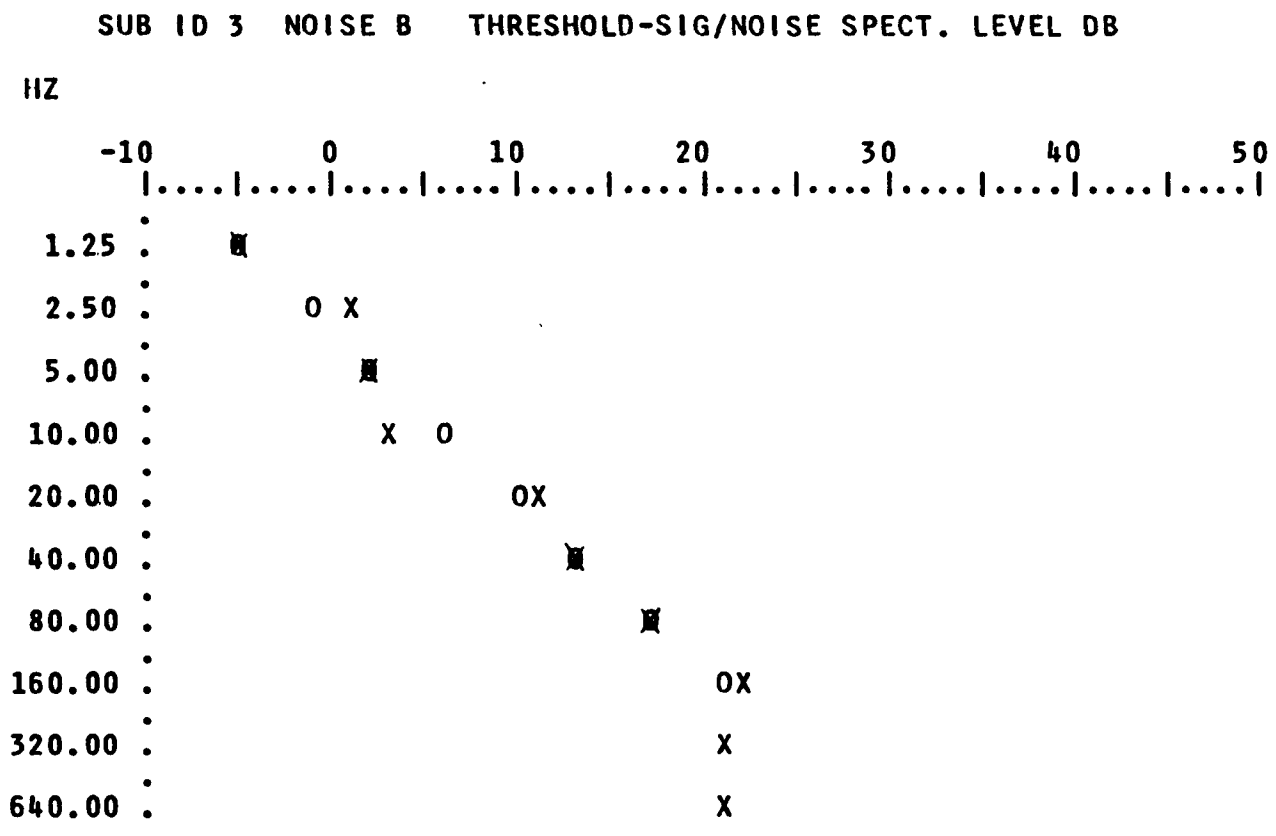

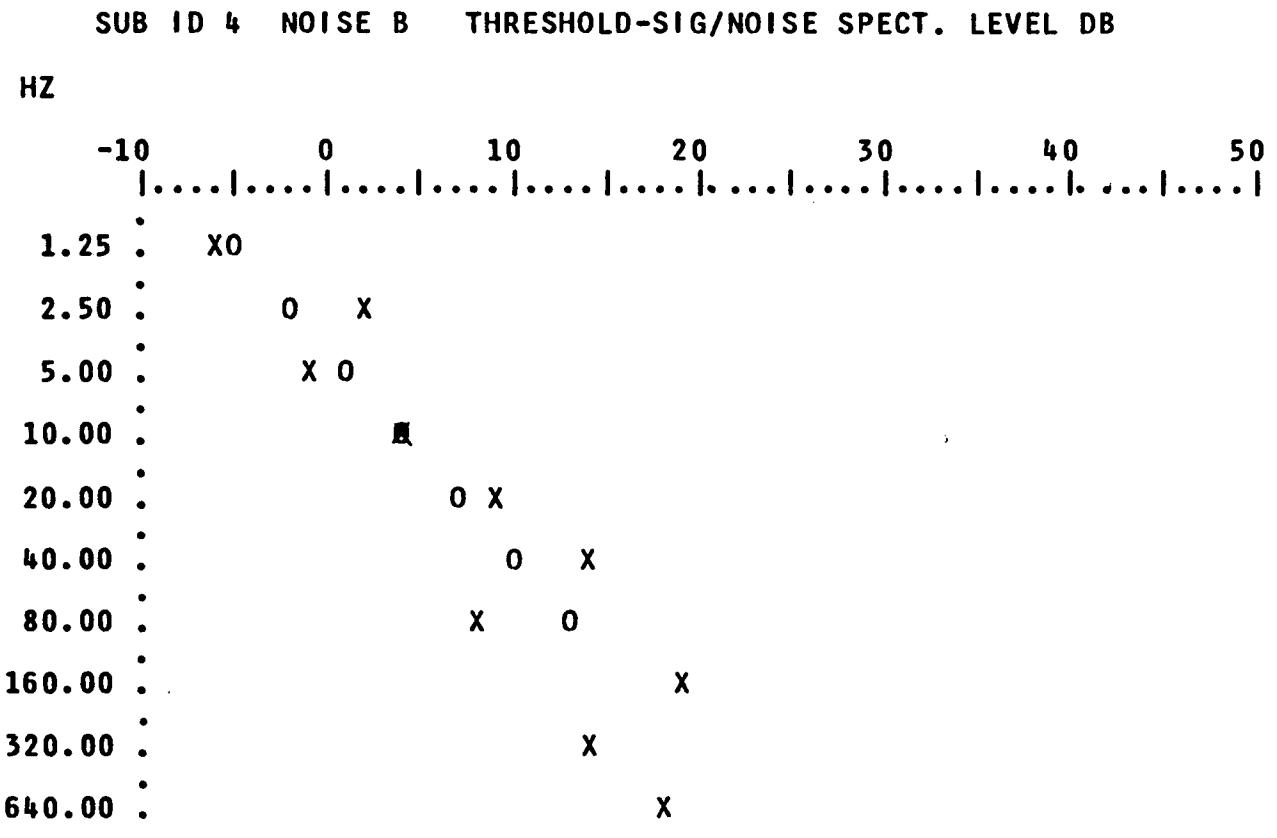
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HZ
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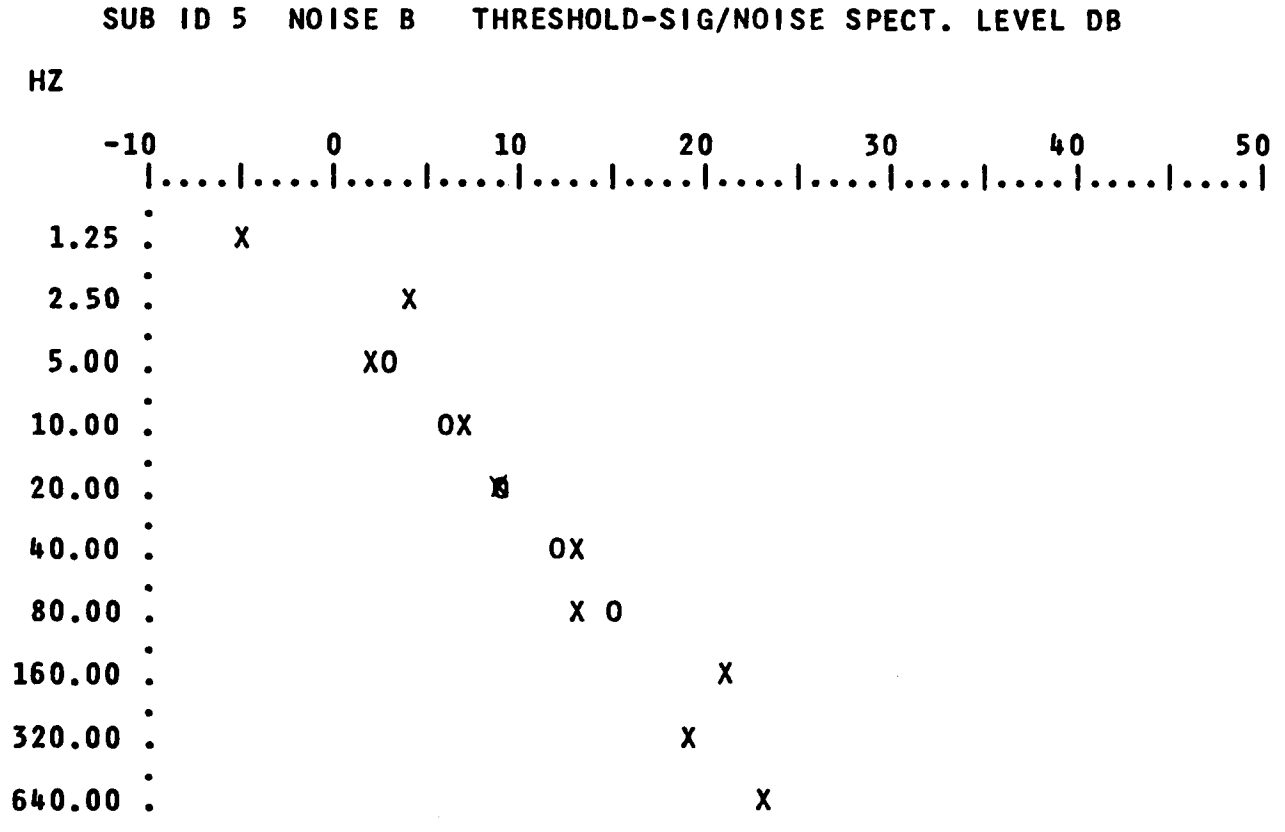
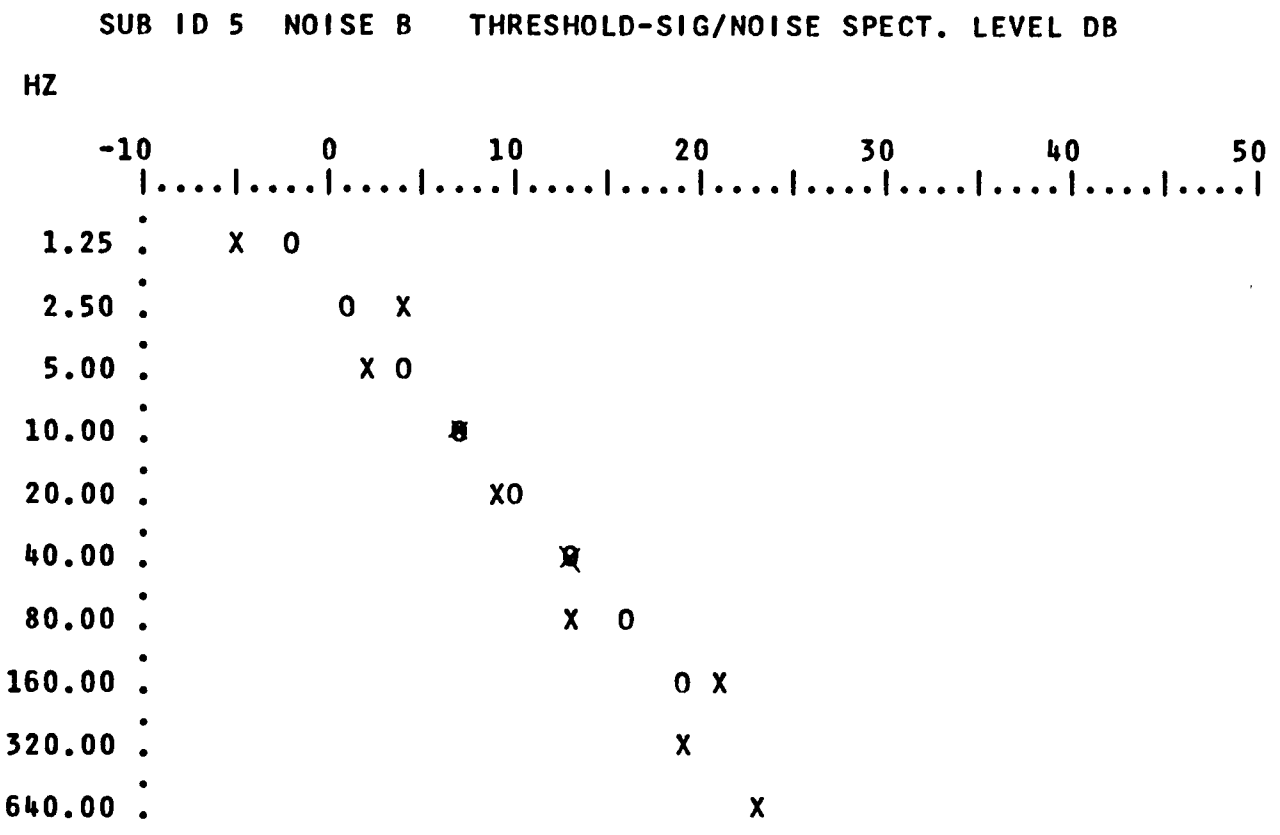
10
20
30
40
50
1.25
2.50
5.00
10.00
20.00
40.00
80.00
160.00
320.00
640.00

SUB ID 5 NOISE B THRESHOLD-SIG/NOISE SPECT. LEVEL DB

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HZ
-10
0
10
20
30
40
50
1.25
2.50
5.00
10.00
20.00
40.00
80.00
160.00
320.00
640.00

critical region by either a 0th or a 1st order polynomial. This means that the function cannot be fitted by a straight line of any slope. All subjects examined, generated curvilinear polynomials of either 2nd or 3rd order over the noise bandwidth range of 5 to 80 or 160 Hz. Figures 45 to 48 display the Noise A threshold functions together with the polynomial sections. All four have a humped shape over the range 5 to 80 Hz showing a rise from 5 to 20 Hz followed by a tendency to flatten off or fall.

C. Outlier Thresholds

Four threshold signal to noise ratios, one at 10 Hz for subject 1, a second a 5 Hz for subject 3, and two at 10 and 20 Hz for subject 4, appeared to be unusually low. These values, -13.99, -16.62, -17.83, and -17.63 dB respectively, were low compared to both the other results of the main investigation and those reported in the literature as well. Subjects achieving these low thresholds knew they had performed unusually well and made comments following the tests to the effect that they had perceived the pattern of the noise. Subject 1, for example, noted that the noise sounded to her as if it were the name "Penelope" and attributed her unusual performance to a consequent ability to detect changes in the sound of the word. Because of the apparently deviant nature of these results together with the comments indicating that a detection strategy differing from the norm had functioned in these cases, an analysis of these results was made to determine if their treatment as outliers could be justified statistically.

The analytical technique used here, is described by Natrella (1963).

FIGURES 45 - 48: Best Fit Threshold versus Noise Bandwidth Curves for Noise A--

computed by orthogonal polynomial analysis. The abscissa gives the noise bandwidth; the ordinate, the signal intensity to noise spectrum level at threshold. The "X"s give the curves computed from empirical data; the "O"s, the best fit orthogonal polynomial.

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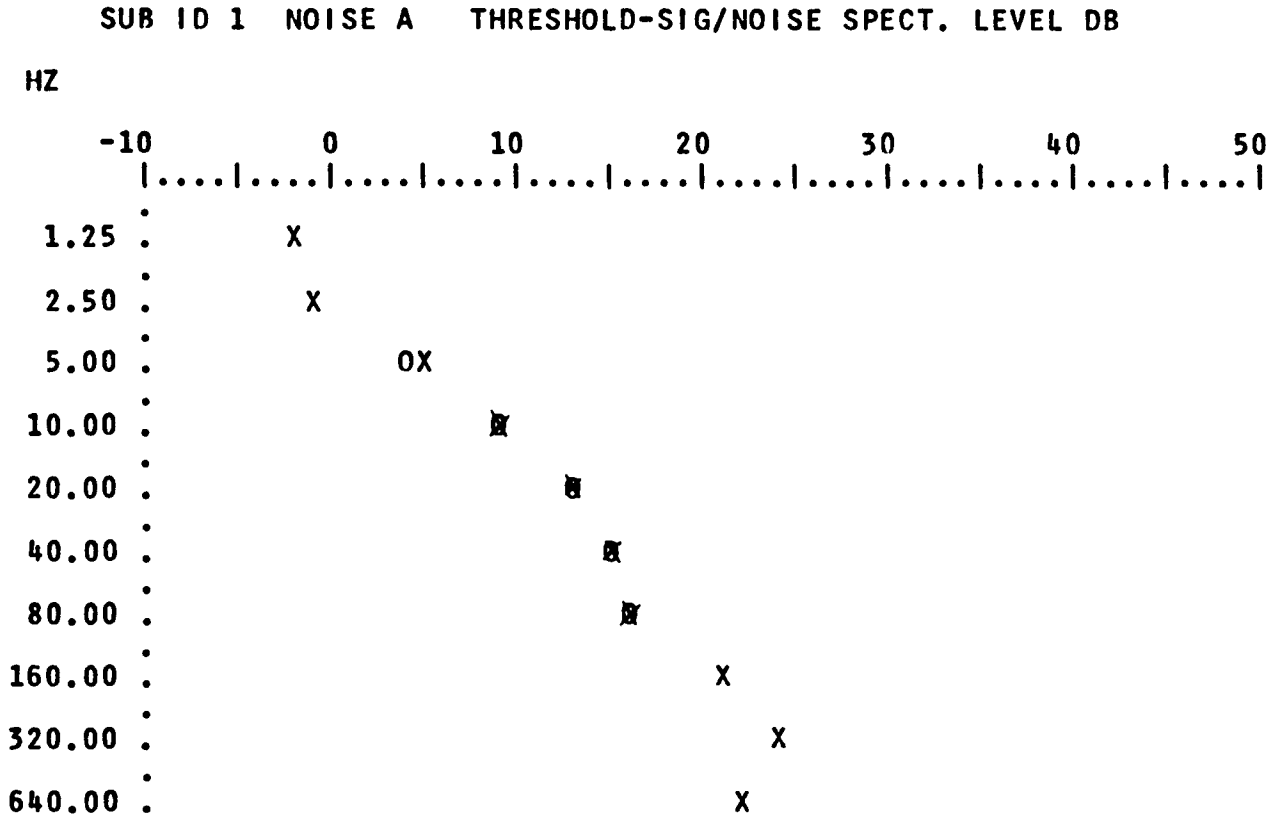

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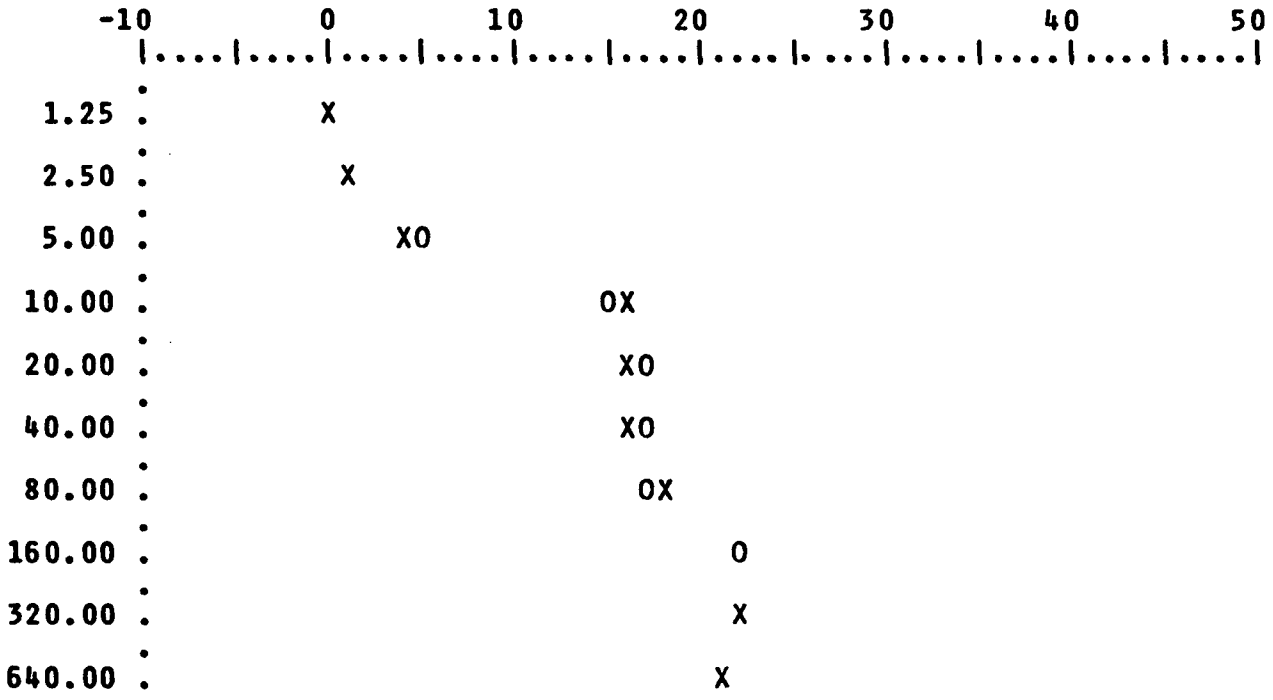

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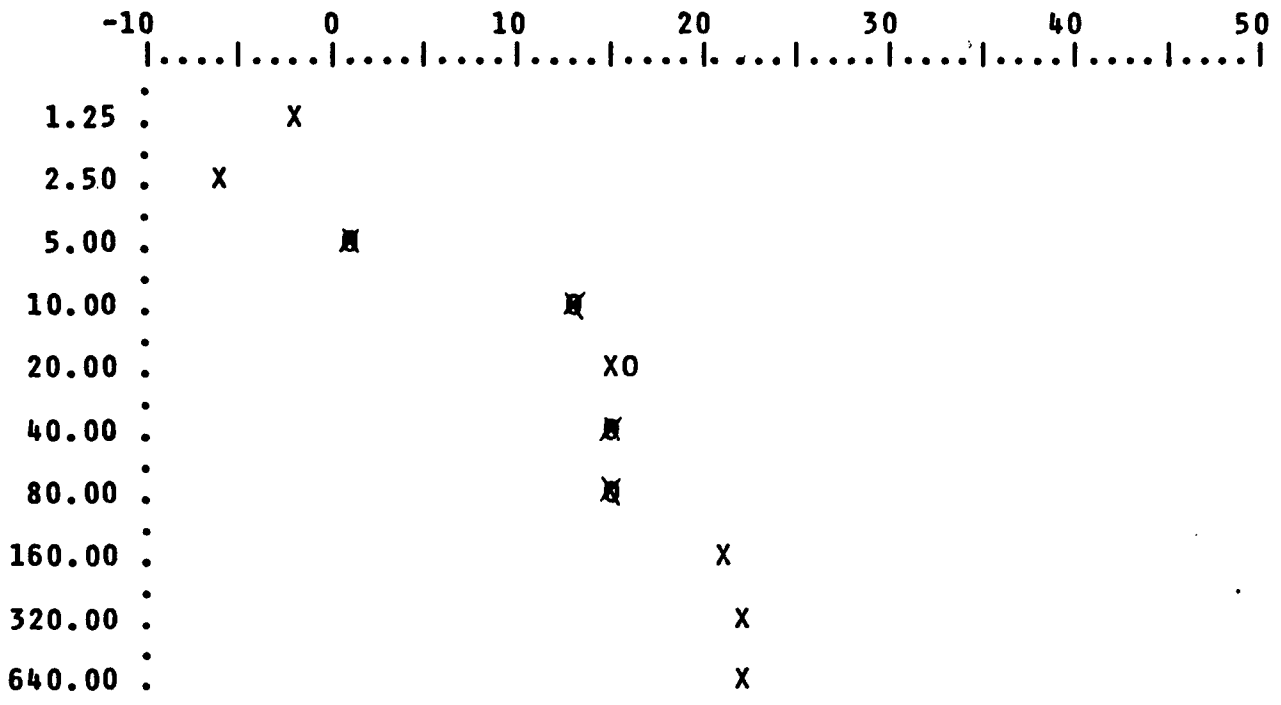

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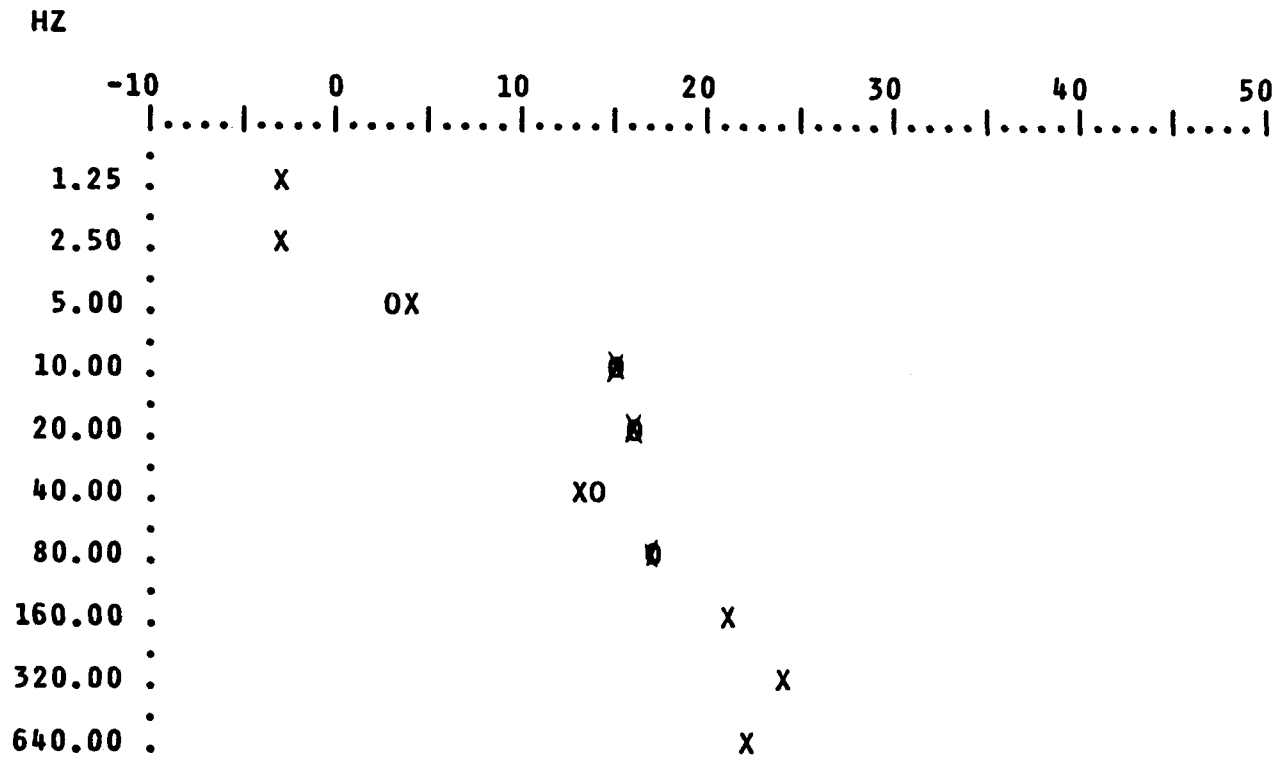


The analysis was conducted for extreme observations in only one direction, namely for lower values. The population mean was assumed to be unknown and the value for the standard deviation was assumed to be unbiased estimate "s" for the set of thresholds minus the outlier values for the subject and noise type in question.

Table 15 displays the results of this analysis. The null hypothesis holding that the extreme values do not differ significantly from the others may be rejected for all four cases at the .01 level. For all other values, in all cases, the null hypothesis is sustained. These results justify, statistically, the subsequent consideration of the results as outlier values. These thresholds however, cannot be dismissed because of this result since they indicate the existence of a unique and sensitive mode of detection which bears further examination.

IV. Discussion of Experimental Results

A. Results in Accord with the Literature

Though the major concern of this study is with the detection of tones in sub-critical bandwidth noise, the noise bandwidths employed did exceed the critical bandwidth. As a result a determination of this bandwidth emerged along with values of the thresholds of tones in supra-critical band noise. In addition, since the narrowest noise bandwidth is virtually a pure tone, the detection of a tone in noise is equivalent to the discrimination of tonal increments so that values for the difference limen could be established. These results are compared below with those generally accepted by the psychoacoustic

TABLE 15

Tests for Outliers

Items	Results		
Subject	1	3	4
Noise	A	B	A
Variance Estimated	2.02	2.4	2.51
Outlier Means	-13.99	-16.62	-17.83 -17.63
Overall Mean	-3.56	-4.86	-7.36
"t" Score Outliers	5.14	4.9	3.77 4.1
"t" Score Next Highest	1.7	1.47	1.49
"t(.01)"	2.88	2.88	2.83
"t(.05)"	2.39	2.39	2.33
Noise Bandwidth	10.00	5.00	10.00 20.00

discipline.

Critical Bandwidths

Though, as noted by de Boer (1962), the comparing of critical bandwidths arising from the various investigations is impeded by the difficulties associated with the gradual sub- to supra-critical transition. For experiments using a pure tone surrounded by masking noise, or for a tone masked by two surrounding tones, a figure, generally accepted as correct at 800 Hz is 150 Hz (Hamilton, 1957; Scharf, 1959, 1961; Greenwood, 1961a; den Brink, 1964; Green, 1965; Canahl, 1971). Experiments based upon critical ratios give about .4 this width but their results depend critically upon the assumed energy ratio at threshold between the signal and the noise (Hawkins and Stevens 1950). Schafer, Gales, Shewmaker, and Thompson (1950) also report a 65 Hz critical band at 800 Hz center frequency, but their transition is so broad that it includes the 150 Hz bandwidth as well.

The broad and ill-defined character of the critical band borders diminishes the strength of the demonstration that all of the subjects in the main investigation showed critical bandwidths near 150 Hz, but the consistency of the results is worth noting. The critical bands were found for Noise B, by finding the intersection of the sub-critical straight line with a horizontal line drawn through those higher bandwidths which appeared to be supra-critical. The bandwidths for Noise A were determined in a similar manner except that the curvilinear sub-critical curves were drawn by eye. More precision did not seem warranted in view of approximate nature of the determination of critical

bandwidths made by previous investigators.

For all subjects, except subject 5, the critical bandwidths for both noises coincided. Subject 5 gave a Noise A critical band which was slightly narrower than the Noise B band. Two bandwidths are given below for each case, a bandwidth "F" which is measured as described above and a bandwidth "3 dB" measured at a point along the sub-critical curve, 3 dB below "F". Subjects 1, 3 and 4 gave "F" bandwidth equal to 200 Hz and "3 dB" bandwidths of 160 Hz for both noises. Subject 5 gave a slightly larger bandwidth for Noise B, 250 and 180 Hz for bandwidths "F" and "3 dB" respectively, but gave the same result as subjects 1, 3 and 4 for noise A. Subject 2 gave narrower bandwidths, 160 Hz for bandwidth "F" and 80 Hz for Bandwidth 3 dB for both noises.

The special conditions of this investigation, frozen noise and controlled presentation parameters, will, in the ensuing discussion, be shown to have little effect in the region immediately below the critical band and for all bandwidths above. Therefore, the critical bandwidths arising from this investigation would be expected to be similar to those reported by other investigators.

Difference Limens

For Noise B, a good estimate of the intensive difference limen may be made from the orthogonal polynomial fitting the sub-critical data since this is a straight line which is horizontal on a signal to noise intensity ratio (not signal to noise spectrum level) versus noise bandwidth display. Since no indication is given by the data that

this line will change slope abruptly, the coefficient of the "0th" order polynomial gives a good estimate of the thresholds at the theoretical pure tone, "0" Hz bandwidth. They are -5.5, -4.6, -6.4 and -4.5 dB for subjects 1, 3, 4 and 5 respectively using the orthogonal polynomial fitting the range 0 to 160 Hz (subject 2's data for Noise B, it will be recalled could not be fitted with a straight line).

The same technique cannot be used for Noise A since its best fitting sub-critical polynomial is of order 2 or higher. However, in this case, since the phase relationship between the tone and the 1.25 Hz bandwidth noise is maintained throughout the presentation, this threshold value may be taken as a good estimate of the results of a discrimination test. These thresholds are -4.1, -3.9, -4.6, and -5.0 dB respectively for subjects 1, 3, 4, and 5, respectively. (Subject 2 will be discussed separately).

For the purposes of this estimation, both sets of results have been combined and averaged since this has been the practice in the handling of data against which these results are to be compared. The resulting average, -4.8 dB must be converted from a signal to noise ratio, signal in phase quadrature to the noise, to an intensive difference limen in which the signal increment is in phase with the base tone. This calculation gives a DL of 1.4 dB.

Intensive DL's are sensitive to measurement techniques varying, for example, from .5 dB to 2.3 dB at frequency levels similar to those used in this investigation. (Dimmick and Olson, 1941). However, Harris

(1952) in addition to finding the low DL reported above for loudness memory under optimum conditions in which the subject could listen to the presentation pairs as often as desired before voting, also investigated the DL for conditions approximating those of this main investigation. As noted previously, he used trials comprising pairs of 1000 Hz pulses, 40 msec. rise time, 600 msec. long, and separated by silent periods of .6 sec. This is similar enough to the conditions of the main investigation, which used 800 Hz, 40 msec. rise times, 400 msec. long, and silent periods of .6 sec., to permit comparison.

At a base signal level of 65 dB SPL, Harris found a DL of 1.3 dB which is close to the 1.4 dB found, at 75 dB SPL, in the main investigation.

Wide Band Results

At supra-critical bandwidths, the subject's task in this investigation becomes the detection of a pure tone in wide band noise. There is general agreement about threshold values in this region even for those investigators who disagree about sub-critical values.

Hamilton (1957) testing at a signal frequency of 800 Hz and a duration of 200 msec. reported wide band threshold signal to noise spectrum ratios of approximately 21 dB. Den Brink (1964) testing at the same signal frequency and duration, compared his results to Hamilton's and reported no discrepancies but, unfortunately, neglected to define the reference level of his ordinate. Bos and de Boer (1966) for a center frequency of 1000 Hz and a signal duration of 250 msec. reported thresholds of approximately 19 dB. Other investigators

Fletcher (1940) and Schaffer et al (1950) collected their data at longer signal durations so that some adjustment must be made to permit comparison with the results of the main investigation. They reported 16 and 17 dB respectively (note all values are signal to noise spectrum ratios.)

The signal to noise spectrum levels at threshold for the main investigation range from 20 to 21 dB for Noise A thresholds and from 18 to 21 for Noise B thresholds giving a Noise A average of 21 dB and a Noise B average of 20 dB. These results conform closely to those reported in the literature.

B. New Results

The orthogonal polynomials which give the best fit to the threshold-noise bandwidth data, indicate that significant differences exist between the effects of Noise A and Noise B within the 5 or 10 to 40 Hz noise bandwidth region. In this region, except for subject 2, Noise B produces threshold data fitted best by a straight line with a slope of 3 dB per octave bandwidth which, if taken alone, seems to support Fletcher's contention that energy, or intensity, ratios are the primary perhaps sole, detection factors. However in this same region, Noise A generates threshold data which cannot be fitted by a straight line, the data indicating a significant degradation of subject's performance. Since both noises have virtually identical intensity levels, as averaged over each presentation, a case is made for the intrusion of factors in addition to the mean intensity level of the presentation, which act to influence detection.

Clues to the nature of these factors have been provided by previous investigators as noted earlier. They include the effects which cause subjects to alter criteria, den Brink (1964); envelope fluctuations which confuse and mislead, Bos and de Boer (1965) and de Boer (1962); noise voltage peaks which mislead the subject, Watson (1964) and signal-like noise stimulus features, Pfafflin and Matthews (1966). Since all of them suggest a role for specific features of the noise stimulus, an inspection of the noise envelopes is indicated. (See Figures 50-69). This search for potentially significant features of the noise stimulus is facilitated in the case of this investigation by the fact that the variation of the presentation mean intensity, an apparently significant factor, has been eliminated.

The search for significant stimulus features has been guided by a number of suppositions derived, in part, from the comments of the investigators discussed above, and from the subjective experiences reported by the subjects participating in the main investigation. Note no suggestion is intended, here, that such experiences provide valid psychoacoustic data but merely an indication as to where to look for such data.

Subjects often felt that they had used more than one detection mode, the various modes corresponding roughly, and not always consistently, with particular noise bandwidth regions. All subjects except subject 2 (the only naive listener) reported listening most attentively for changes in the presentation at the time of expected signal onset. For supra-critical bandwidths, most subjects reported listening

for a change in presentation quality at expected signal onset time, this quality change being associated with the presence of a signal. For sub-critical bandwidths most subjects reported listening for changes in level most of the time but occasionally detecting changes in the quality of the noise fluctuations in the presence of a signal for the wider sub-critical bands. Three subjects claimed to have occasionally perceived the noise pattern with great clarity within the 5 to 40 Hz region and to have based their vote on minute changes in this pattern. All subjects admitted to basing their decisions on changes in overall loudness when the other modes could not be applied with confidence and Subject 2 reported relying solely on this last mode throughout the experiment.

Since the most significant differences between the effects of the two noises occured in the 5 to 40 Hz region, the dominant mode, listening for changes in the region of signal onset provided a guide in the search for potential factors This search resulted in the uncovering of two factors, which seemed to account for the differences in the two threshold curves and suggested the following hypothesis.

An Hypothesis Concerning Sub-Critical Detection

The proposed hypothesis holds that given presentations comprising gated tones masked by gated noise of sub-critical bandwidth in which the time of expected signal onset is known to the subject, the resulting thresholds may be affected by two factors acting to cause significant deviations around a base-line threshold established by the presentation-wide tone-to-noise intensity ratios. Since each of these two

factors occur on a probabilistic basis, they may or may not be present in any single presentation. If they do occur mutually, one may act either to assist or oppose the effects of the other.

Factor 1 is the Local Intensity Mean (LIM) of the noise taken over a section of the noise coincident with the onset period (on-transition time) of the tone and acts either to degrade or enhance detection within the critical period in which the subject is listening for the onset of the tone. Deviations of the LIM in this period above the presentation mean will degrade detection while deviations below this mean will enhance detection. The extent of the threshold change is related to the extent of the deviation.

Factor 2 is a result of variations in the noise envelope similar to that produced by the addition of a tone signal to the noise (signal-like variations of the noise envelope) which occur during the expected onset period of the signal and act either to mislead or confuse the subject so as to degrade his threshold. If the subject is misled, he will report the presence of a signal when it is actually absent; if he is confused, he may react by raising his criteria. The influence of this factor depends upon how closely the noise envelope excursion mimics the effects of adding a signal to the noise.

Regions of Influence of the Factors

Because of the specific conditions imposed upon the tone and noise parameters in this study, each of the two factors, when present, are most likely to exert maximum influence over specific sub-critical bandwidth regions.

For Factor 1, this region lies between 10 to 40 Hz noise bandwidths and is a result of holding constant the duration and mean intensity of each presentation. This produces two opposing limitations on the LIMs which are least restricting within the 10 to 40 Hz region.

One limitation on the extent of the deviation of the LIM, arises from restrictions imposed on the maximum allowable rate of change of noise envelopes by narrow bandwidths combined with relatively short presentation durations. As the noise bandwidth increases, however, the envelope can change more rapidly and larger deviations from the presentation mean may occur within the duration of the presentation.

This action is, however, opposed by the increased number of independent bits of information concerning the noise waveform, encompassed within the 40 msec signal onset time, as the noise bandwidth increases (sampling theorem; Black, 1953). As a result, the extent of the deviation of the LIM about its presentation-wide mean decreases in accordance with the statistical law governing the standard error of the mean as a function of sample size. For sampling periods of 40 msec , the smallest sample size, one independent value, and hence the largest LIM deviation, is most likely near a noise bandwidth of 13 HZ or below.

The combined effects of both restrictions determine the spectral position of the Factor 1 maxima.

Factor 2, signal-like variations in the noise envelope, also depends heavily upon the rate of envelope variation permitted by the processing filter but the spectral region in which the noise envelope is most likely to mimic the effects of adding a signal can only be

estimated approximately because of the effects of the complexity of the input waveform to the filter. Though the bandwidth required for the filter output to reach maximum in 40 msec. in response to an abrupt change in input is easily found to be 19.5 Hz for the filter used in this study, complex changes in input levels, can produce signal-like variations in the envelope of the output noise at lower bandwidths. In addition, since bandwidths merely limit the maximum rate of change of the output noise, wider bandwidths may also generate signal-like envelope changes in the noise. A bandwidth range of 5 to 40 Hz probably will include the region in which Factor 2 effects are most likely to be greatest.

Evidence Supporting the Hypothesis

Factor 1.--An analysis of the local intensity means for segments of the both noises, of all durations, at all locations within a presentation, and for all bandwidths, supported the expectations of the hypothesis. This analysis, conducted by computer, explored all possibilities exhaustively, displaying the results as in Fig. 49 (local intensity mean to presentation-wide spectrum level versus noise bandwidth). The results were examined for trends similar to those found for the threshold curves; specifically for Noise A and B curves which, for Noise A, followed the curved sub-critical path of the Noise A thresholds, and, for Noise B, followed a straight line with a slope of approximately 3 dB per octave. No assumptions were imposed upon this search, LIM's from segments of all durations and positions within the presentation, were examined.

FIGURE 49: Local Intensity Average in the Region Surrounding the Signal Onset Versus Noise Bandwidth

The abscissa gives the noise bandwidth; the ordinate, the local noise intensity referred to the overall mean spectrum level of the presentation. The "X"s represent Noise A, the "O"s, Noise B.

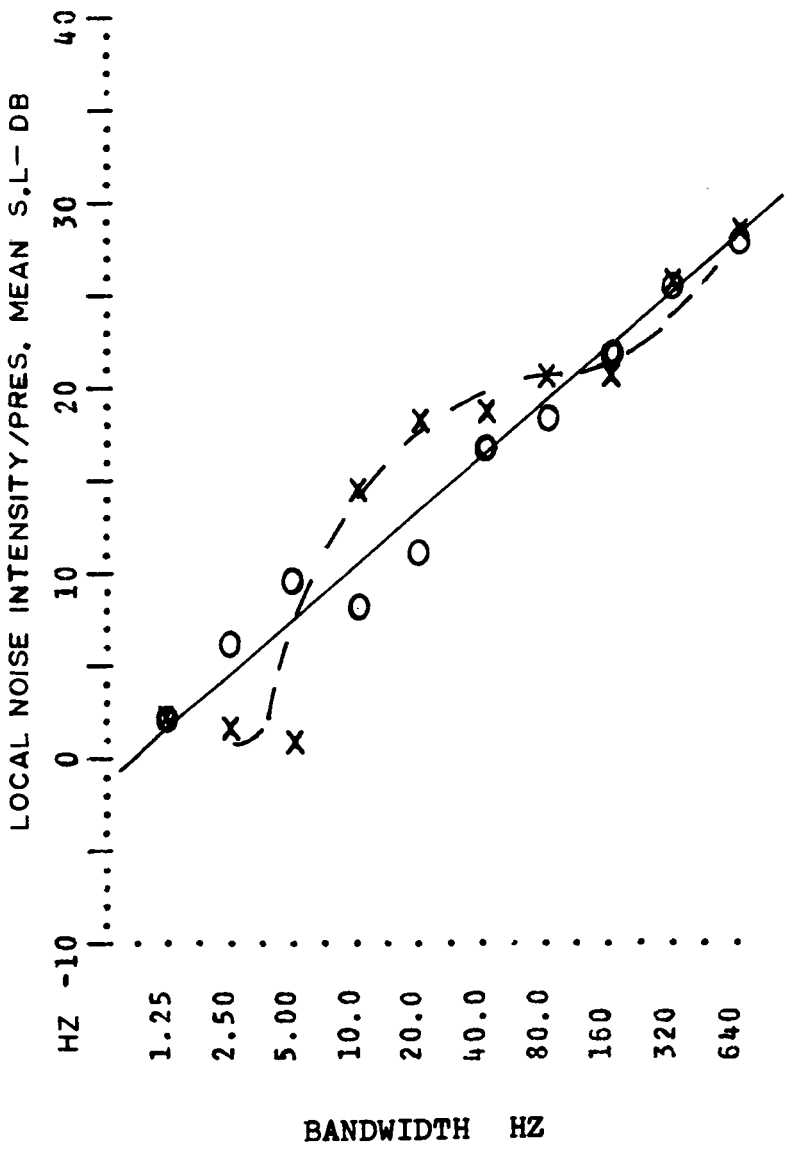
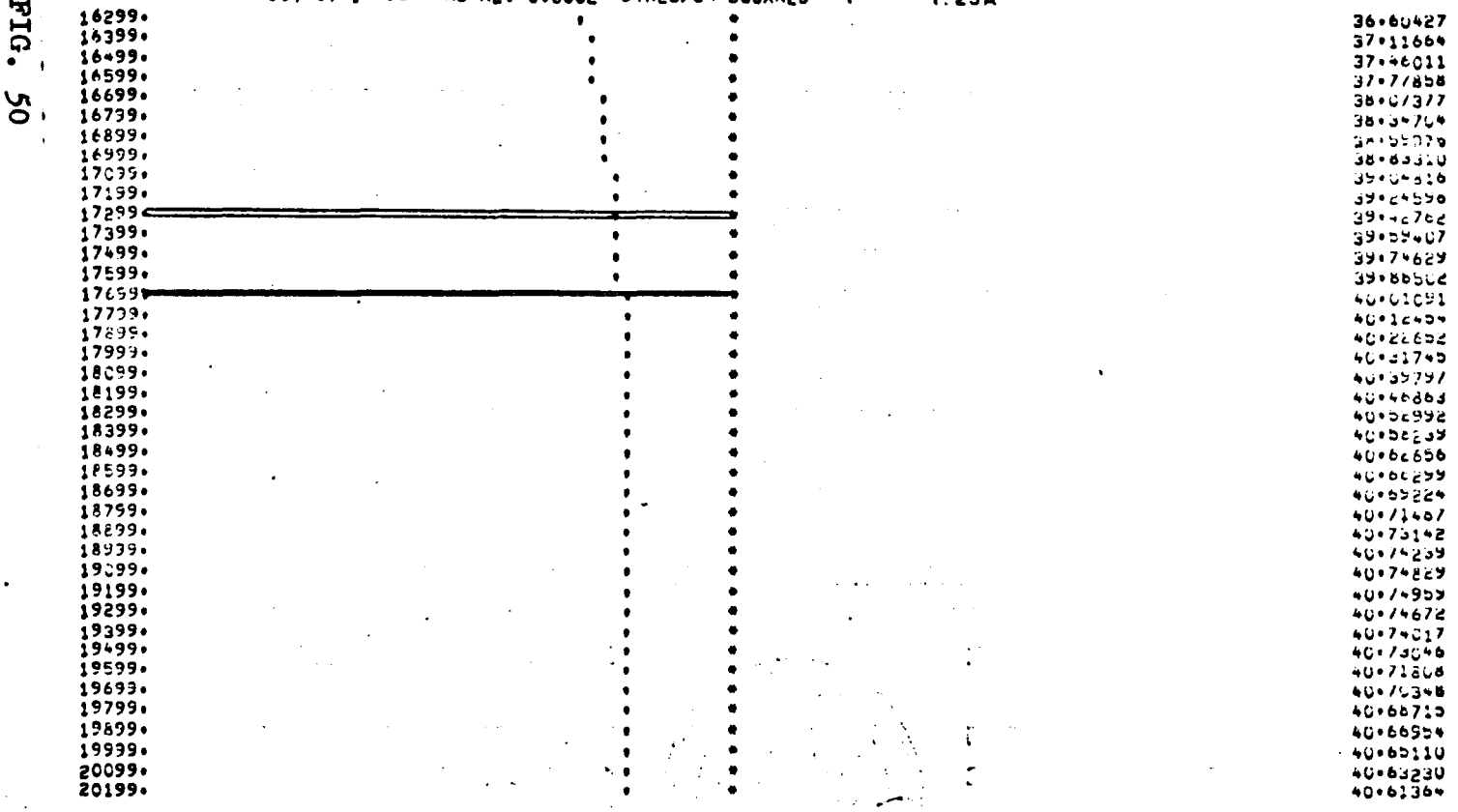

LOCAL NOISE INTENSITY/PRES. MEAN S.L- DB
HZ -10
0
10
20
30
40
1.25
2.50
5.00
10.0
20.0
40.0
80.0
160
320
640
BANDWIDTH HZ

FIGURE 50-69: Envelopes of Noise Presentations

The abscissa, left column, gives the computer cycles in increments of 100 cycles (10 msec.); the ordinate, right column, gives the mean RMS value in dB re: 0.0002 (SPL) of the noise for the corresponding 10 msec. interval. The two double ruled lines enclose the signal onset transition period. The envelopes are shown prior to being shaped by rise and fall transitions.

OUTPUT IN DB RMS RE. 0.0002 DYNES/CM SQUARED 1.25A



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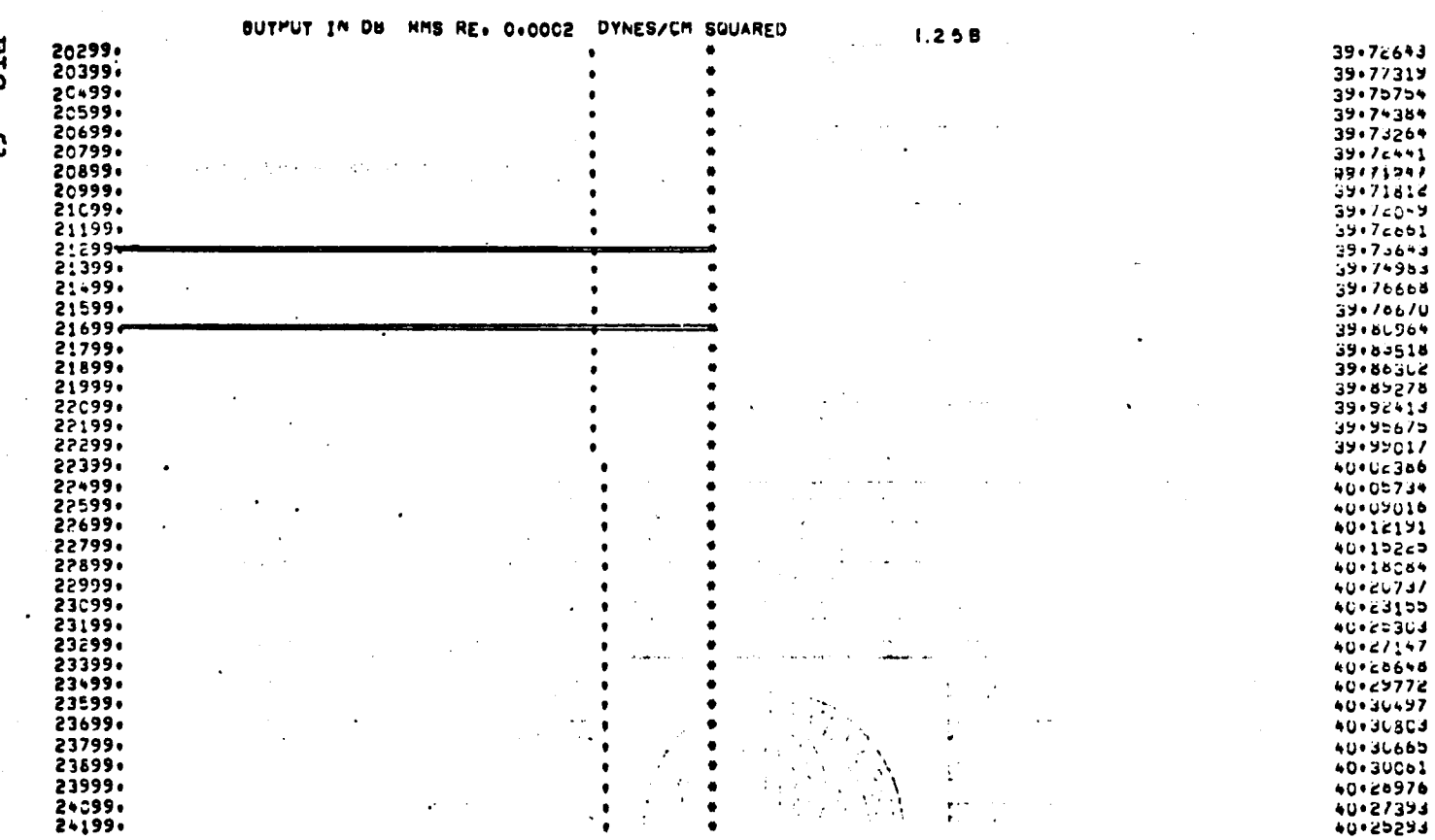
TRIAL NUMBER C HAS BEEN TAPED

FIG. 50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1.25 B
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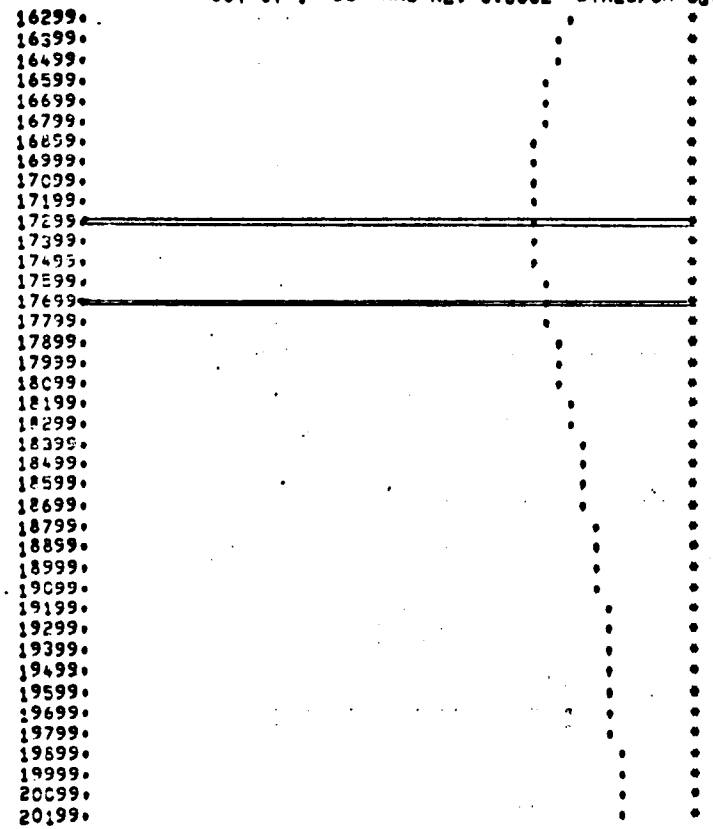
20499. 39.75754
20599. 39.74384
20699. 39.73264
20799. 39.72441
20899. 39.71947
20999. 39.71816
21099. 39.72059
21199. 39.72661
21299. 39.73643
21399. 39.74983
21499. 39.76668
21599. 39.78670
21699. 39.80964
21799. 39.83518
21899. 39.86302
21999. 39.89278
22099. 39.92413
22199. 39.95675
22299. 39.99017
22399. 40.02386
22499. 40.05734
22599. 40.09016
22699. 40.12191
22799. 40.15229
22899. 40.18084
22999. 40.20737
23099. 40.23155
23199. 40.25303
23299. 40.27147
23399. 40.28648
23499. 40.29772
23599. 40.30497
23699. 40.30803
23799. 40.30665
23899. 40.30061
23999. 40.28976
24099. 40.27393
24199. 40.25293
MEAN=DB= .9999999963E+00 STANU. DEV. OF MEAN= .2635974812E-01 NORM. DEV.= .2635974862E-01
TRIAL NUMBER C HAS BEEN JAMED

FIG. 51

OUTPUT IN DB RMS RE. 0.0002 DYNES/CM SQUARED 2.5 A



16299.	39.14877
16399.	38.78933
16499.	38.33298
16599.	37.89816
16699.	37.49528
16799.	37.13562
16899.	38.[illegible]
16999.	36.59814
17099.	36.44564
17199.	36.38479
17299.	36.42312
17399.	36.55906
17499.	36.76062
17599.	37.07032
17699.	37.40756
17799.	37.77415
17899.	38.15580
17999.	38.54175
18099.	38.92472
18199.	39.29987
18299.	39.66387
18399.	40.01311
18499.	40.34457
18599.	40.65633
18699.	40.94678
18799.	41.21443
18899.	41.45942
18999.	41.66332
19099.	41.88801
19199.	42.07500
19299.	42.24462
19399.	42.39823
19499.	42.53885
19599.	42.66994
19699.	42.79359
19799.	42.91048
19899.	43.01956
19999.	43.11942
20099.	43.20845
20199.	43.28473

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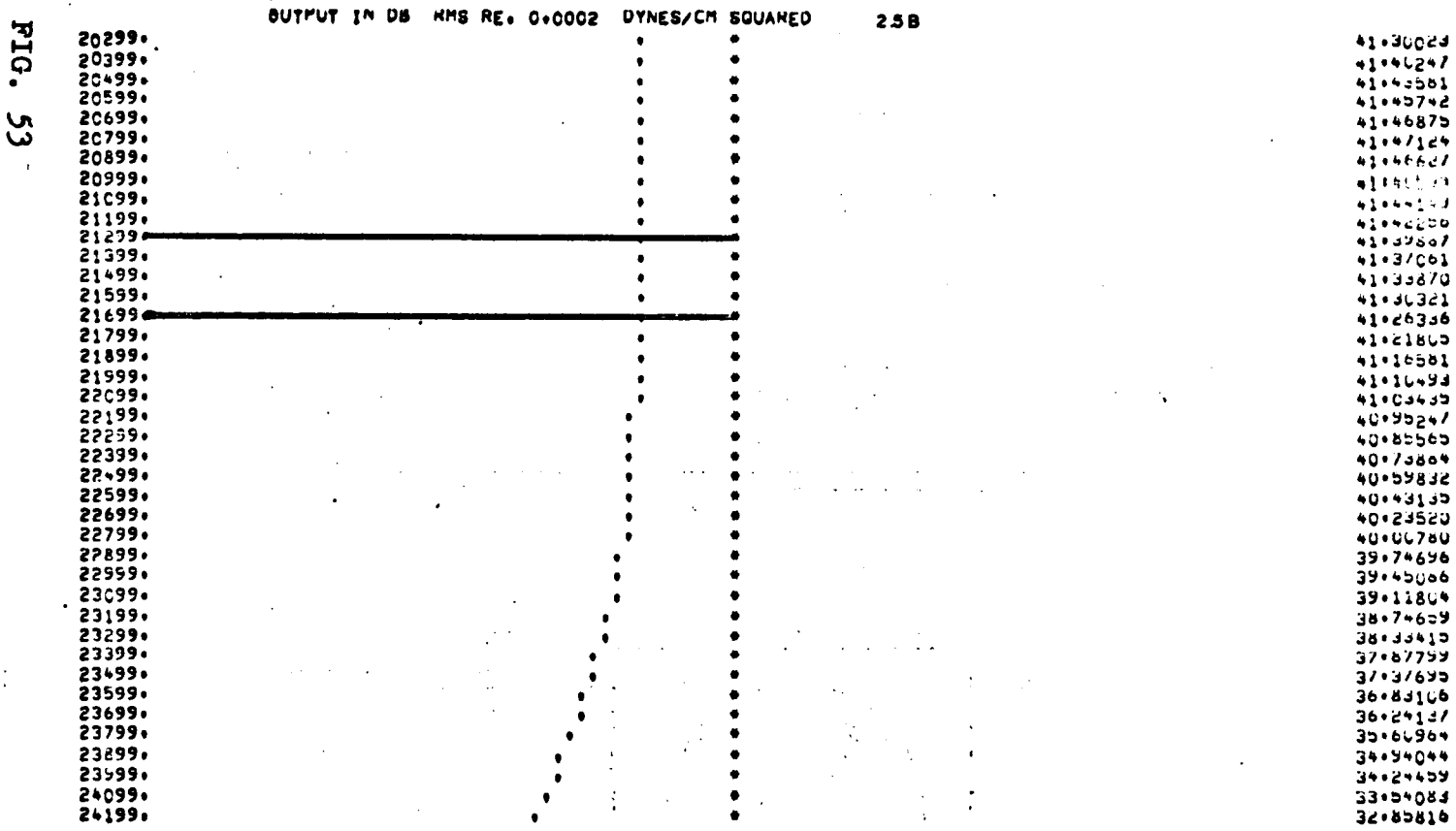
TRIAL NUMBER 0 HAS BEEN TAPED

FIG. 52



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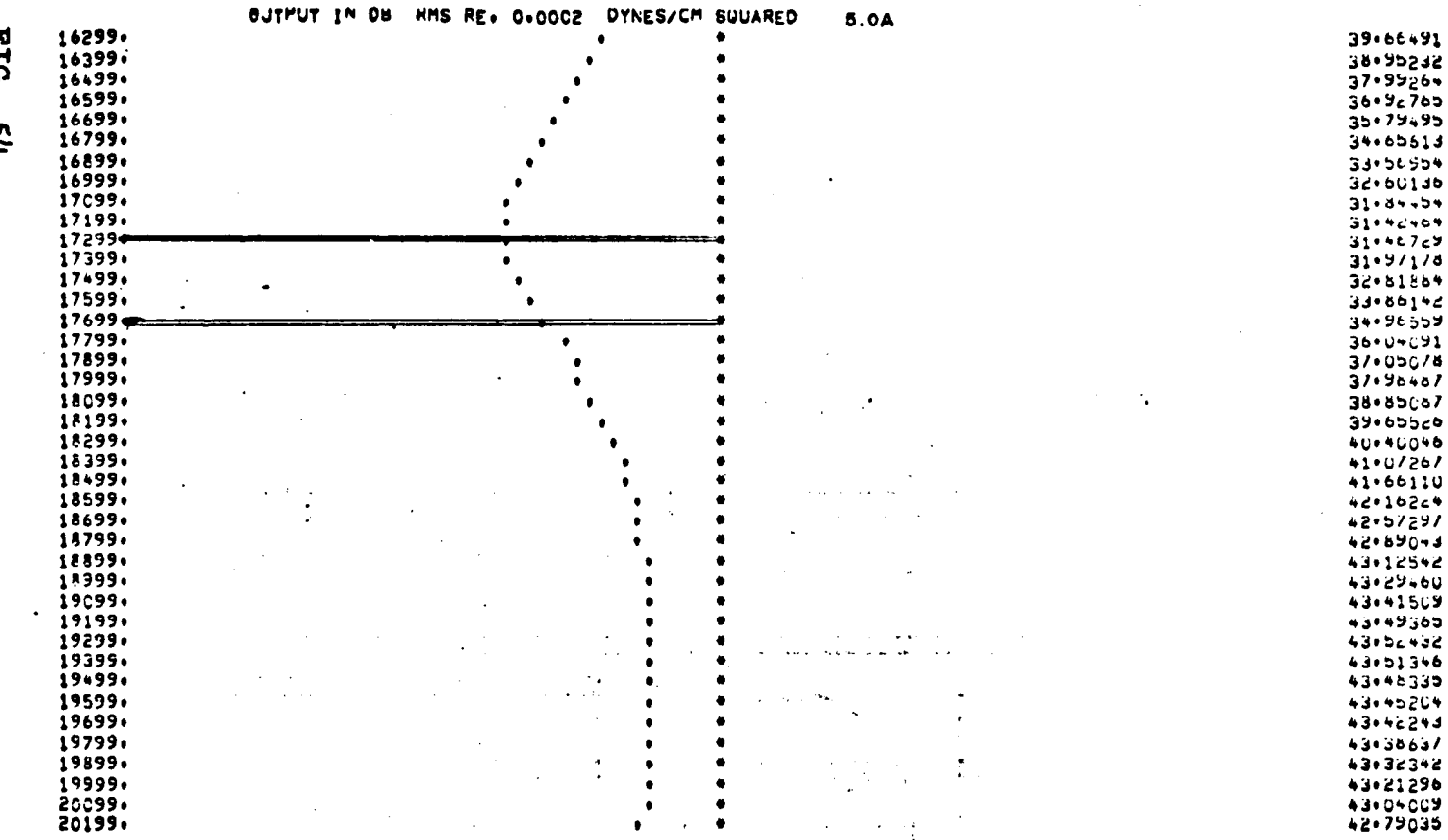
TRIAL NUMBER C HAS BEEN TAPED

FIG. 53

FIG. 54


OUTPUT IN DB RMS RE. 0.0002 DYNES/CM SQUARED 5.0A
16299.
16399.
16499.
16599.
16699.
16799.
16899.
16999.
17099.
17199.
17299.
17399.
17499.
17599.
17699.
17799.
17899.
17999.
18099.
18199.
18299.
18399.
18499.
18599.
18699.
18799.
18899.
18999.
19099.
19199.
19299.
19399.
19499.
19599.
19699.
19799.
19899.
19999.
20099.
20199.
39.66491
38.95232
37.95264
36.92765
35.79495
34.65613
33.56954
32.60136
31.84454
31.42464
31.46729
31.97178
32.81864
33.86142
34.96559
36.04091
37.05078
37.96467
38.85067
39.65526
40.40046
41.07267
41.66110
42.16264
42.57297
42.89043
43.12542
43.29460
43.41509
43.49365
43.52432
43.51346
43.48335
43.45204
43.42243
43.38637
43.32342
43.21296
43.04009
42.79035
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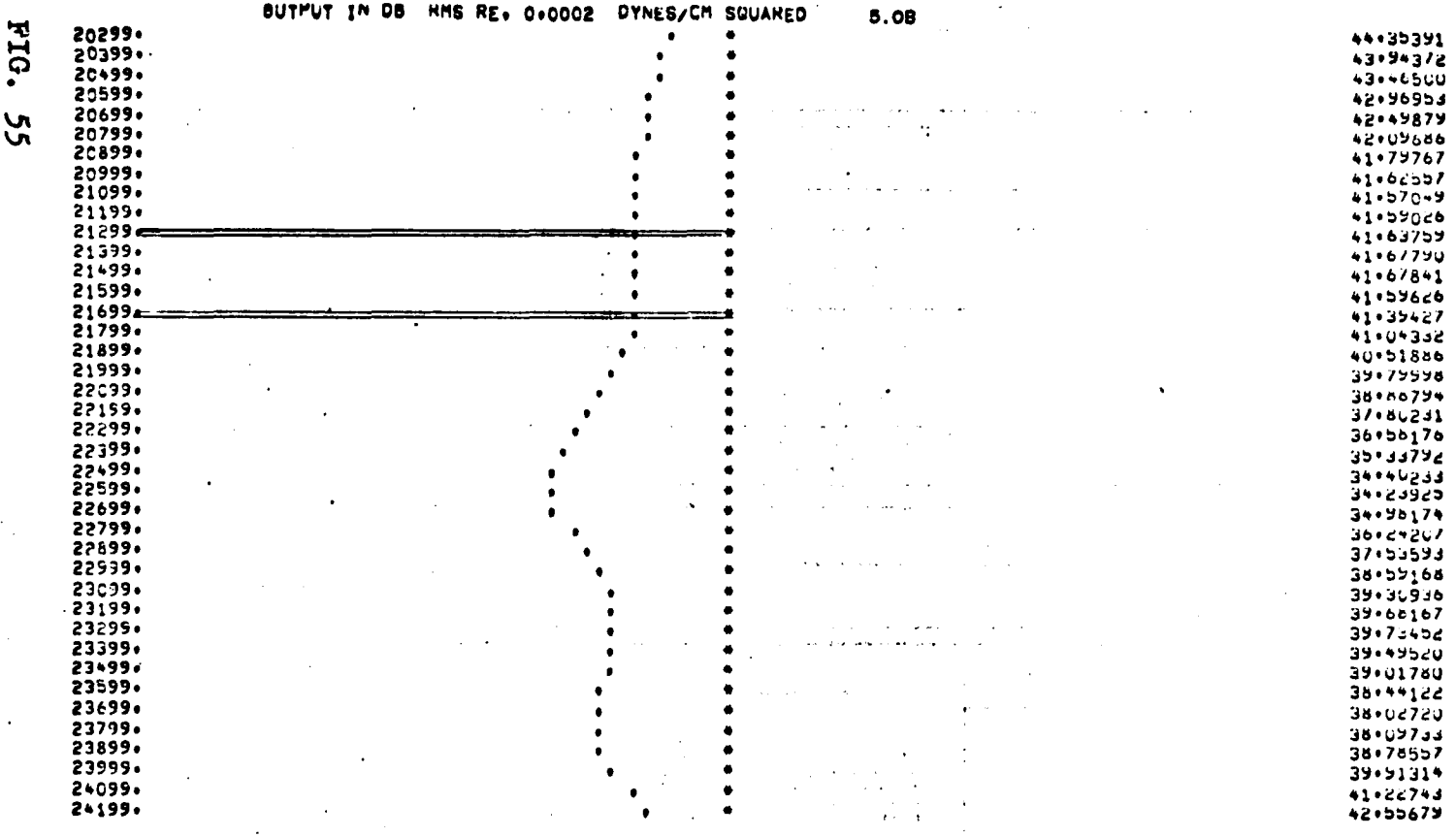
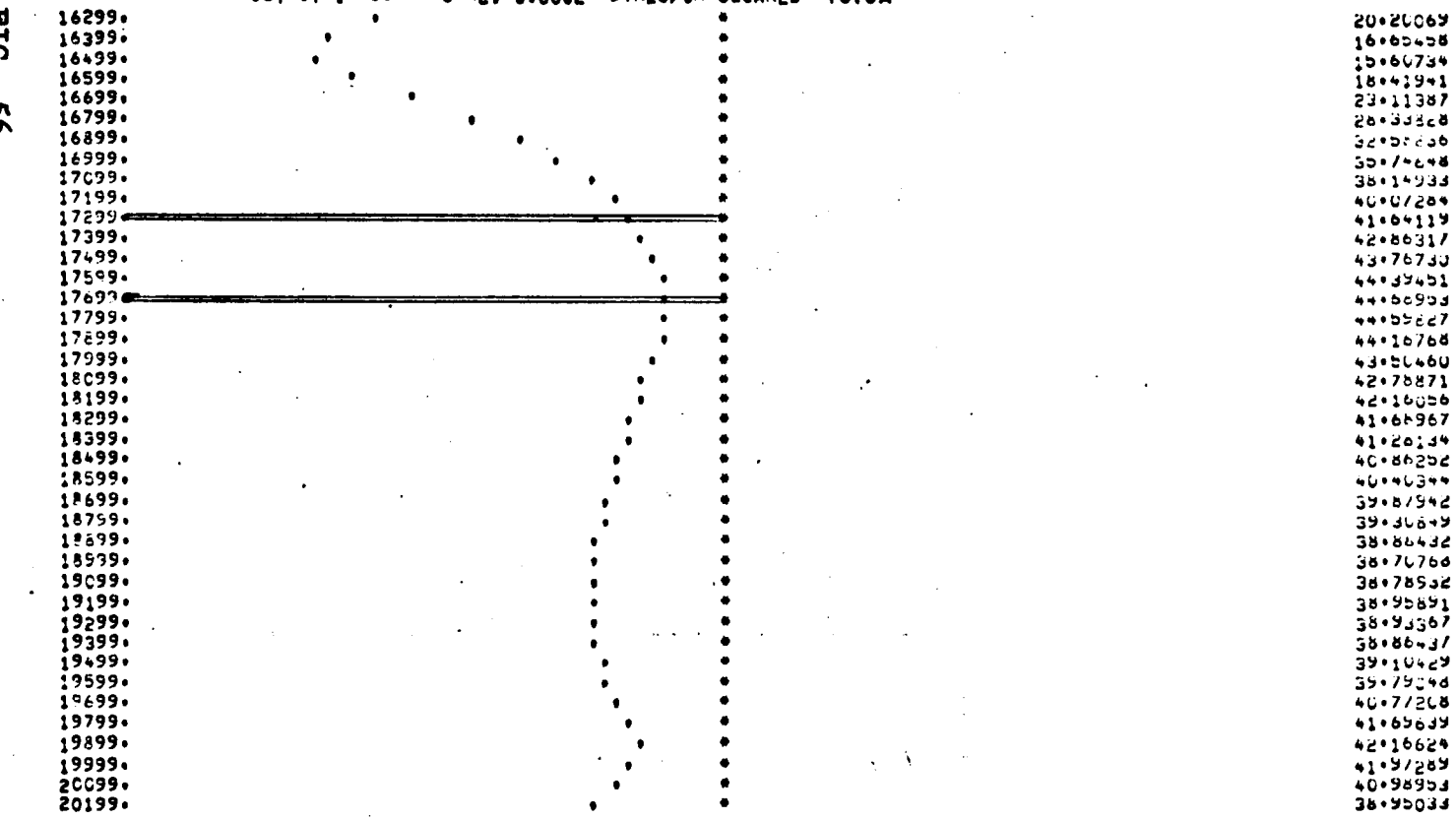

OUTPUT IN DB RMS RE. 0.0002 DYNES/CM SQUARED 5.0B
20299. 44.35391
20399. 43.94372
20499. 43.46500
20599. 42.96953
20699. 42.49879
20799. 42.09686
20899. 41.79767
20999. 41.62557
21099. 41.57049
21199. 41.59026
21299. 41.63759
21399. 41.67790
21499. 41.67841
21599. 41.59626
21699. 41.39427
21799. 41.04332
21899. 40.51886
21999. 39.79598
22099. 38.86794
22199. 37.80231
22299. 36.66176
22399. 35.55792
22499. 34.60233
22599. 34.23929
22699. 34.36174
22799. 36.24207
22899. 37.53593
22999. 38.59168
23099. 39.30936
23199. 39.66167
23299. 39.75452
23399. 39.69520
23499. 39.61780
23599. 39.44122
23699. 39.02720
23799. 38.39733
23899. 38.78557
23999. 39.51314
24099. 41.22743
24199. 42.55679
MEAN=DB= .9999999989E+00 STANU. DEV. OF MEAN= .2663278712E+00 NORM. DEV. = .2663278715E+00
TRIAL NUMBER C HAS BEEN TAKEN

FIG. 55

OUTPUT IN DB RMS RE. 0.0002 DYNES/CM SQUARED 10.0A

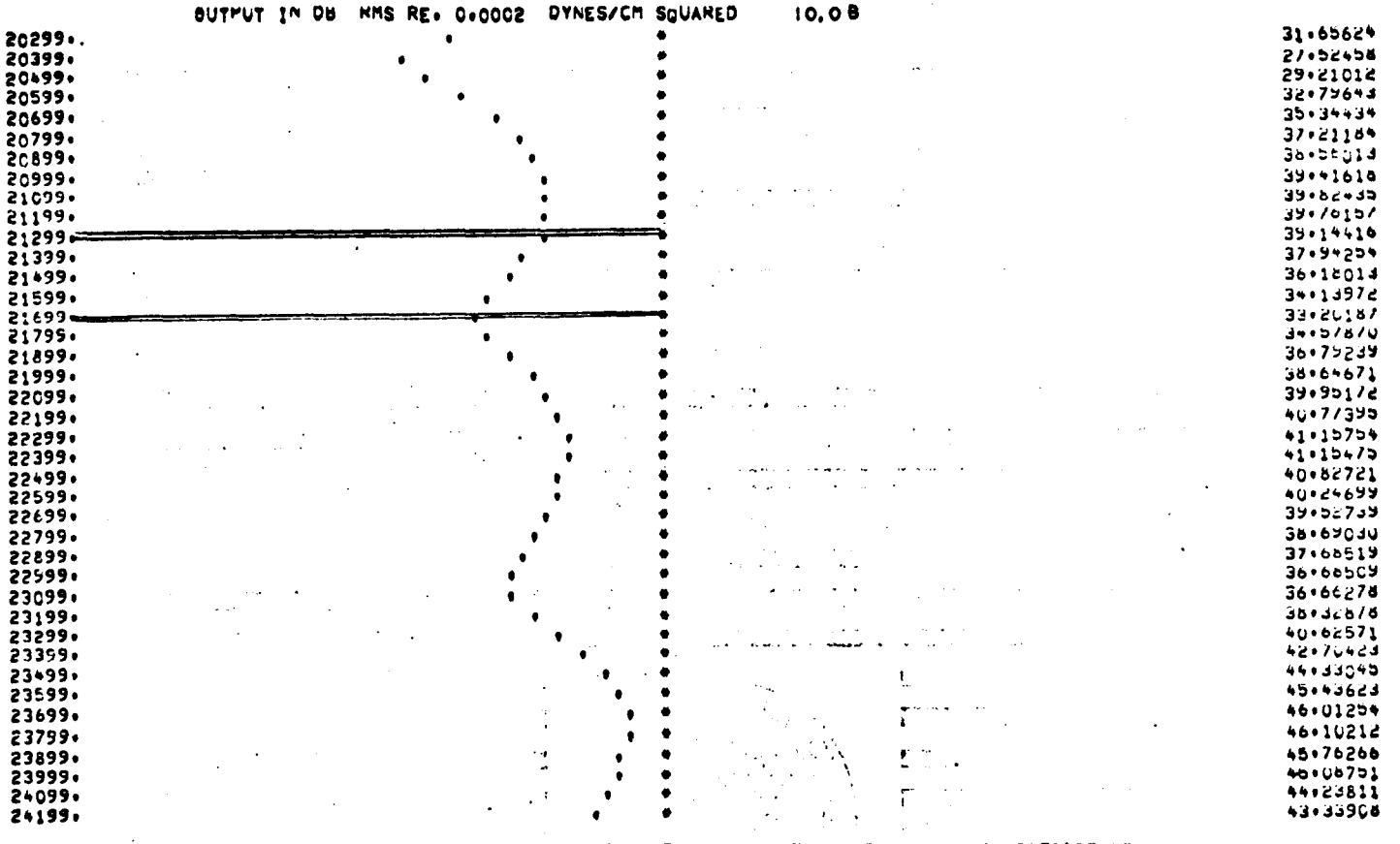


MEAN-DB= .1000000001E+01 STAND. DEV. OF MEAN= .4462914551E+00 NORM. DEV. = .4462914548E+00

TRIAL NUMBER 0 HAS BEEN TAPED

FIG. 56

OUTPUT IN DB RMS RE. 0.0002 DYNES/CM SQUARED 10.0 B



MEAN-DB= .1000000001E+01 STAND. DEV. OF MEAN= .4690947116E+00 NORM. DEV. = .4690947112E+00

TRIAL NUMBER 0 HAS BEEN TAPED

FIG. 57

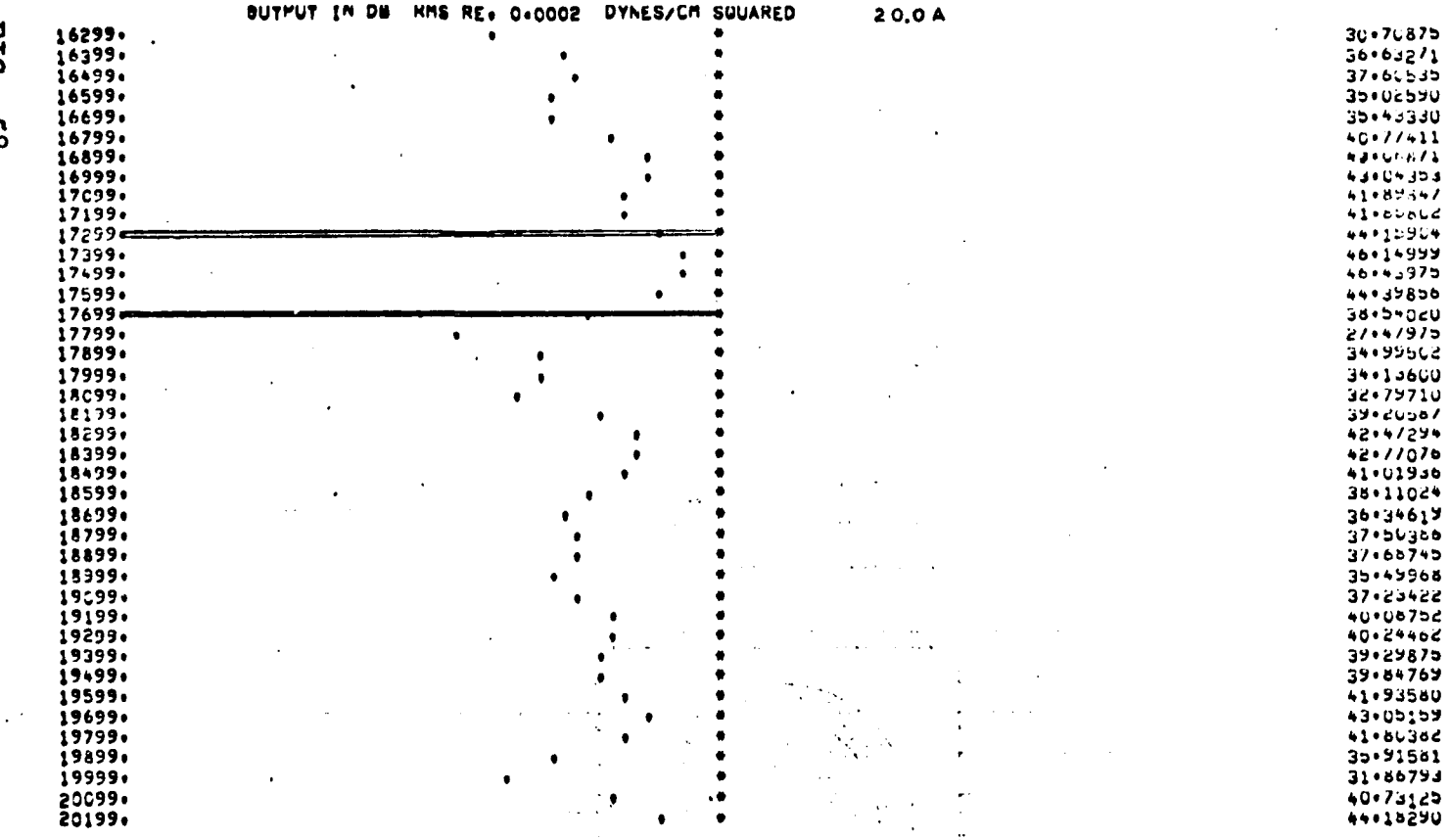


FIG. 58

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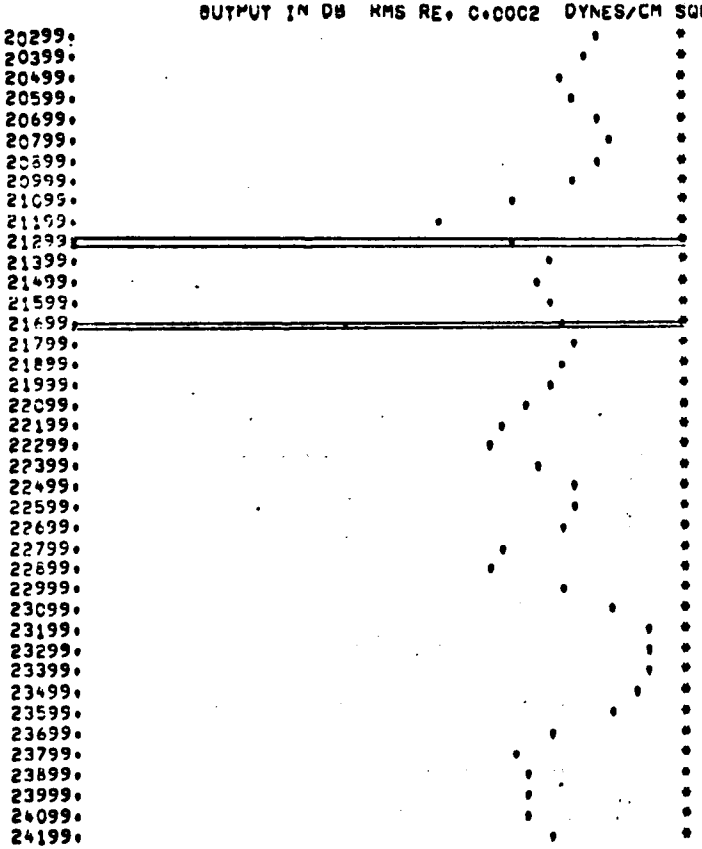
OUTPUT IN DB RMS RE. C.0002 DYNES/CM SQUARED 20.0B



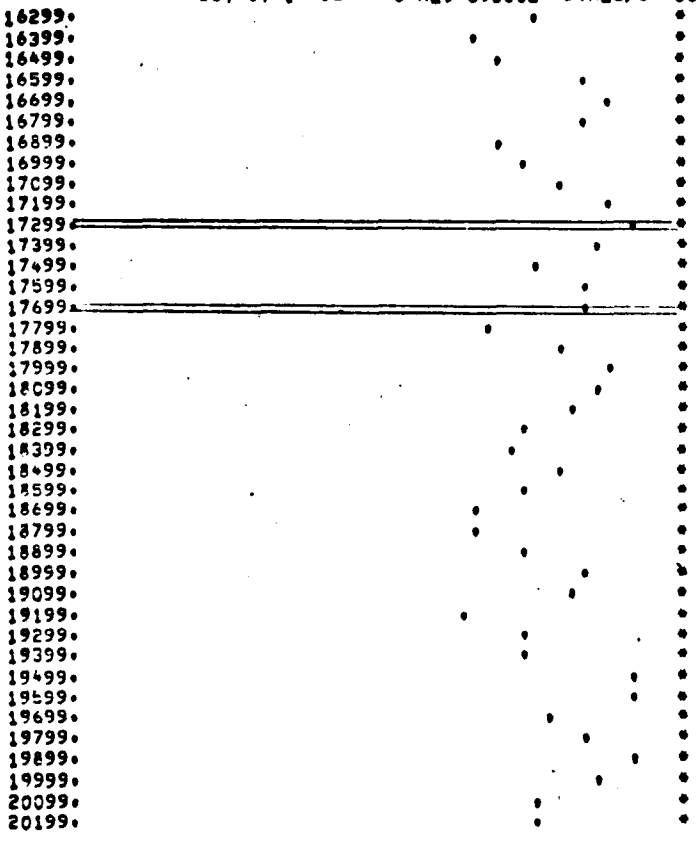
20299.	42.73772
20399.	41.69660
20499.	39.06378
20599.	40.23357
20699.	42.97316
20799.	43.88209
20899.	[illegible]
20999.	[illegible]
21099.	[illegible]
21199.	29.62228
21299.	35.83120
21399.	38.06655
21499.	37.64972
21599.	38.00724
21699.	39.37686
21799.	40.02922
21899.	39.64731
21999.	38.30641
22099.	36.45872
22199.	34.81273
22299.	33.42609
22399.	37.11113
22499.	40.32121
22599.	40.96898
22699.	39.22682
22799.	34.76708
22899.	33.22460
22999.	39.28834
23099.	43.82999
23199.	46.06438
23299.	46.62696
23399.	46.31014
23499.	45.35453
23599.	43.07909
23699.	38.87430
23799.	35.16145
23899.	36.17062
23999.	36.87865
24099.	36.28922
24199.	38.07737

MEAN-DB= .9999999996E+00 STAND. DEV. OF MEAN= .4617942331E+00 NORM. DEV. = .4617942333E+00

TRIAL NUMBER C HAS BEEN TAPED

FIG. 59

OUTPUT IN DB RMS RE. 0.0002 DYNES/CM SQUARED 40.0A

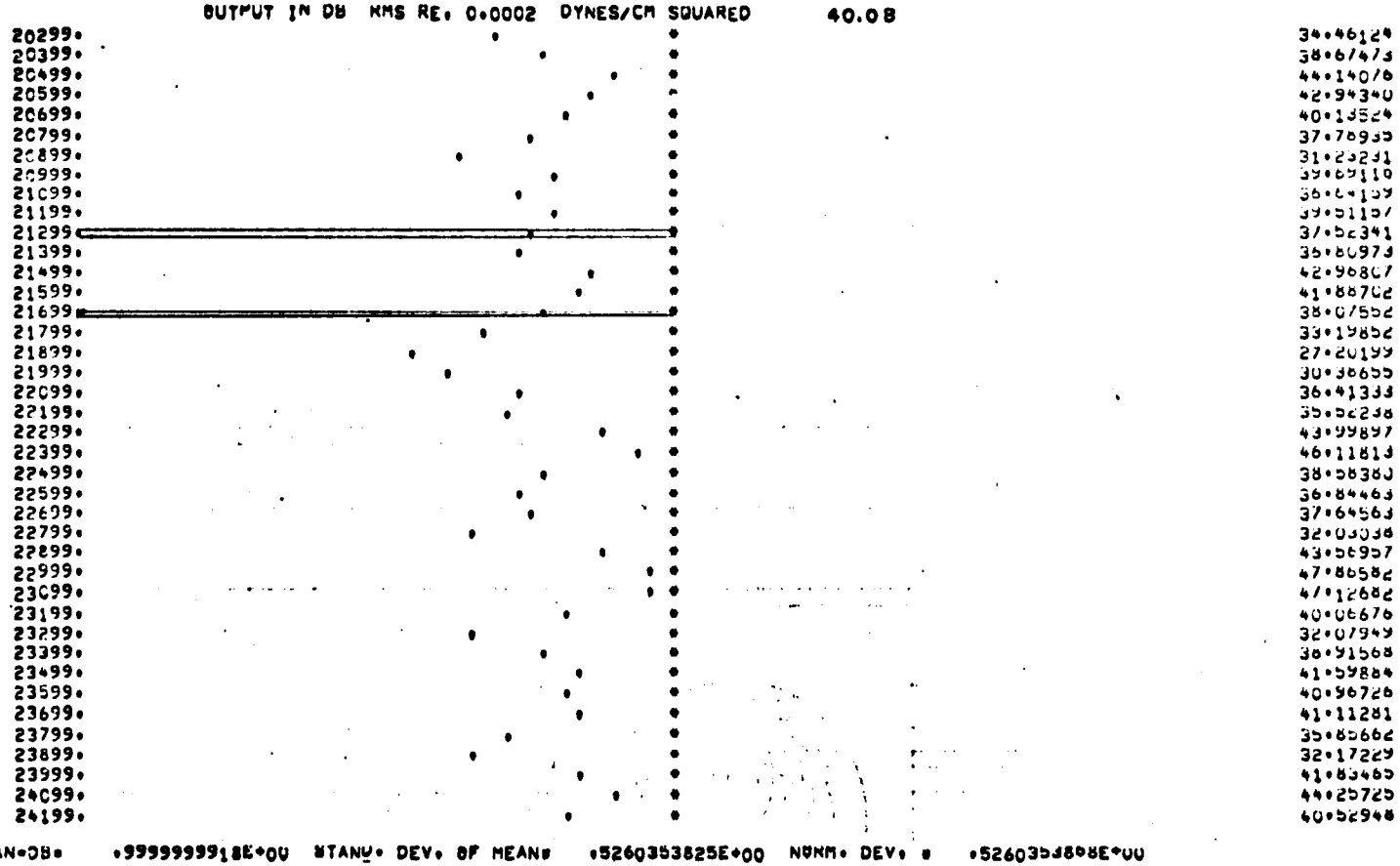


16299.	37.87619
16399.	32.69200
16499.	34.71007
16599.	41.03445
16699.	43.15019
16799.	41.61721
16899.	34.32649
16999.	36.15574
17099.	39.04611
17199.	43.97736
17299.	45.64969
17399.	42.51313
17499.	37.38606
17599.	41.22046
17699.	41.80970
17799.	33.69622
17899.	39.31827
17999.	43.23706
18099.	42.11045
18199.	40.20945
18299.	36.94496
18399.	35.11781
18499.	39.82397
18599.	36.29394
18699.	32.66232
18799.	32.86108
18899.	36.45932
18999.	41.73351
19099.	40.49033
19199.	31.44242
19299.	36.36673
19399.	36.29991
19499.	45.35245
19599.	45.82950
19699.	38.97317
19799.	41.52341
19899.	45.01889
19999.	42.02133
20099.	37.45605
20199.	37.37447

MEAN-DB= .9999999984E+00 STAND. DEV. OF MEAN= .4412061656E+00 NORM. DEV. = .4412061603E+00

TRIAL NUMBER 0 HAS BEEN TAPED

FIG. 60


OUTPUT IN DB RMS RE. 0.0002 DYNES/CM SQUARED 40.08
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20499. 44.14076
20599. 42.94340
20699. 40.13524
20799. 37.76955
20899. 31.25251
20999. 39.69110
21099. 36.64159
21199. 39.51157
21299. 37.52341
21399. 35.80973
21499. 42.96807
21599. 41.86702
21699. 38.07552
21799. 33.19852
21899. 27.20199
21999. 30.38655
22099. 36.41333
22199. 35.52238
22299. 43.99897
22399. 46.11813
22499. 38.58360
22599. 36.84463
22699. 37.64563
22799. 32.03038
22899. 43.56957
22999. 47.86582
23099. 47.12682
23199. 40.06676
23299. 32.07949
23399. 38.91568
23499. 41.59884
23599. 40.96726
23699. 41.11281
23799. 35.85662
23899. 32.17229
23999. 41.85465
24099. 44.25725
24199. 40.52948
MEAN=DB= .99999999 18E+00 STANU. DEV. OF MEAN= .52603538 25E+00 NORM. DEV. = .52603548 08E+00
TRIAL NUMBER C HAS BEEN TAPED

FIG. 61

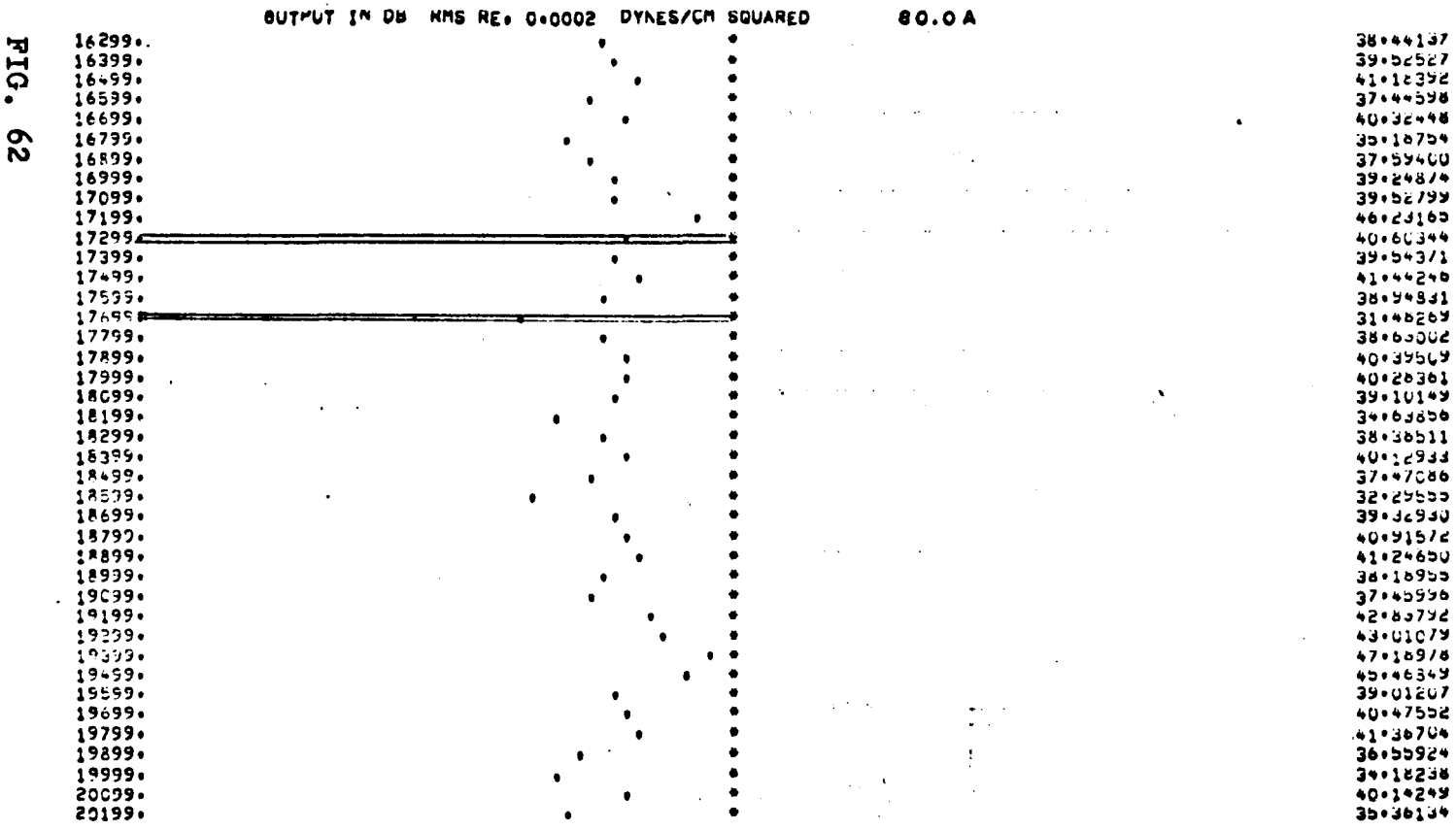

OUTPUT IN DB RMS RE: 0.0002 DYNES/CM SQUARED 80.0 A
16299. 38.44137
16399. 39.52527
16499. 41.12392
16599. 37.44598
16699. 40.32498
16799. 35.18754
16899. 37.59400
16999. 39.24874
17099. 39.52799
17199. 46.23165
17299. 40.60344
17399. 39.54371
17499. 41.44240
17599. 38.94831
17699. 31.46269
17799. 38.65002
17899. 40.39569
17999. 40.28361
18099. 39.10149
18199. 34.69856
18299. 38.38611
18399. 40.12933
18499. 37.47086
18599. 32.29555
18699. 39.32930
18799. 40.91572
18899. 41.24650
18999. 38.18955
19099. 37.45996
19199. 42.85792
19299. 43.01079
19399. 47.16978
19499. 45.46349
19599. 39.01207
19699. 40.47552
19799. 41.38704
19899. 36.55924
19999. 34.18238
20099. 40.14249
20199. 35.36134
MEAN-DB= .9999999998E+00 STAND. DEV. OF MEAN= .3923405032E+00 NORM. DEV. = .3923405032E+00
TRIAL NUMBER C HAS BEEN TAPED

FIG. 62

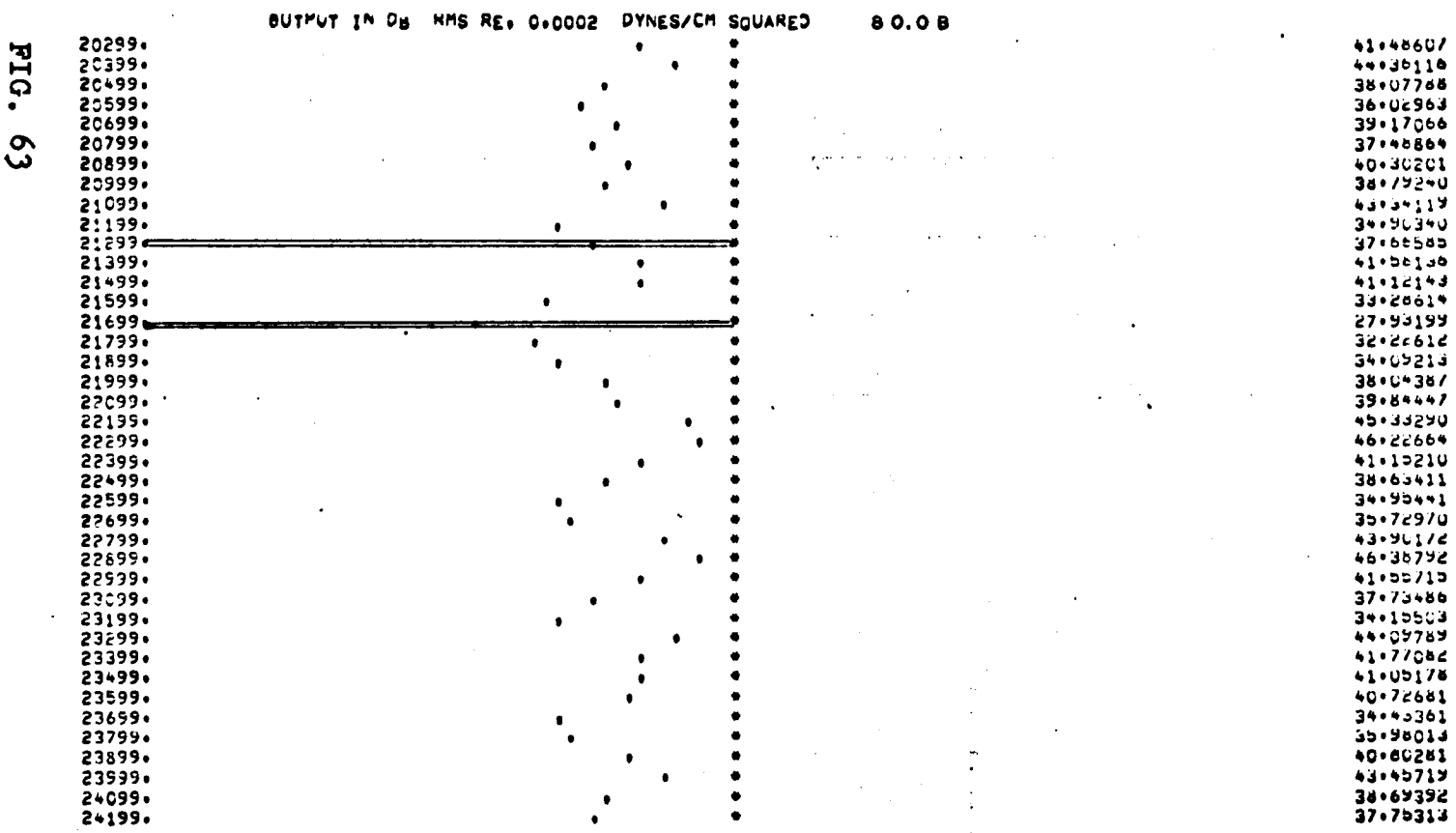

OUTPUT IN DB RMS RE: 0.0002 DYNES/CM SQUARED 80.0 B
20299. 41.48607
20399. 44.36118
20499. 38.07768
20599. 36.02963
20699. 39.17066
20799. 37.48864
20899. 40.30201
20999. 38.79240
21099. 43.34119
21199. 34.90340
21299. 37.66585
21399. 41.56136
21499. 41.12143
21599. 33.26614
21699. 27.93199
21799. 32.26616
21899. 34.09213
21999. 38.04387
22099. 39.84447
22199. 45.33290
22299. 46.22664
22399. 41.19210
22499. 38.63411
22599. 34.95441
22699. 35.72970
22799. 43.90172
22899. 46.38792
22999. 41.55715
23099. 37.73486
23199. 34.15503
23299. 44.09789
23399. 41.77082
23499. 41.05178
23599. 40.72681
23699. 34.45361
23799. 35.98013
23899. 40.80281
23999. 43.45719
24099. 38.69392
24199. 37.75313
MEAN=DB= .9999999969E+00 STAND. DEV. OF MEAN= .4532861706E+00 NORM. DEV. = .4532861760E+00
TRIAL NUMBER C HAS BEEN TAPED

FIG. 63

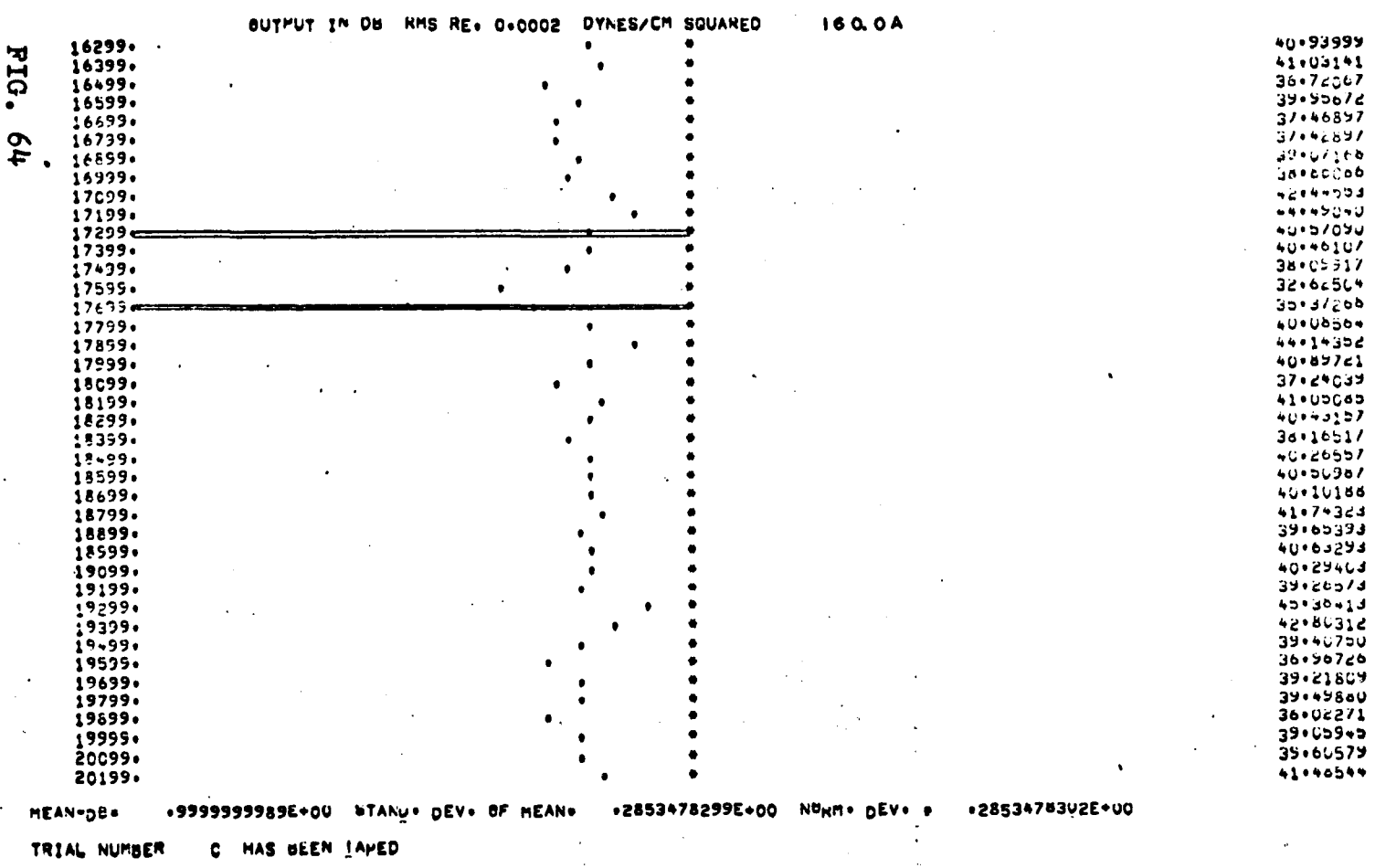

OUTPUT IN DB RMS RE. 0.0002 DYNES/CM SQUARED 160.0A
16299. 40.93999
16399. 41.03141
16499. 36.72067
16599. 39.95672
16699. 37.46897
16799. 37.42897
16899. 39.07166
16999. 38.85066
17099. 42.44753
17199. 44.45040
17299. 40.57090
17399. 40.46107
17499. 38.05917
17599. 32.62504
17699. 35.37268
17799. 40.08564
17899. 44.14352
17999. 40.89761
18099. 37.24039
18199. 41.05065
18299. 40.43157
18399. 38.16517
18499. 40.26557
18599. 40.50987
18699. 40.10188
18799. 41.74324
18899. 39.65393
18999. 40.63294
19099. 40.29403
19199. 39.26579
19299. 45.38413
19399. 42.80312
19499. 39.40750
19599. 36.96726
19699. 39.21809
19799. 39.49880
19899. 36.02271
19999. 39.05945
20099. 35.60579
20199. 41.46544
MEAN-DB= .9999999989E+00 STAND. DEV. OF MEAN= .2853478299E+00 NORM. DEV. = .2853478302E+00
TRIAL NUMBER C HAS BEEN TAPED

FIG. 64

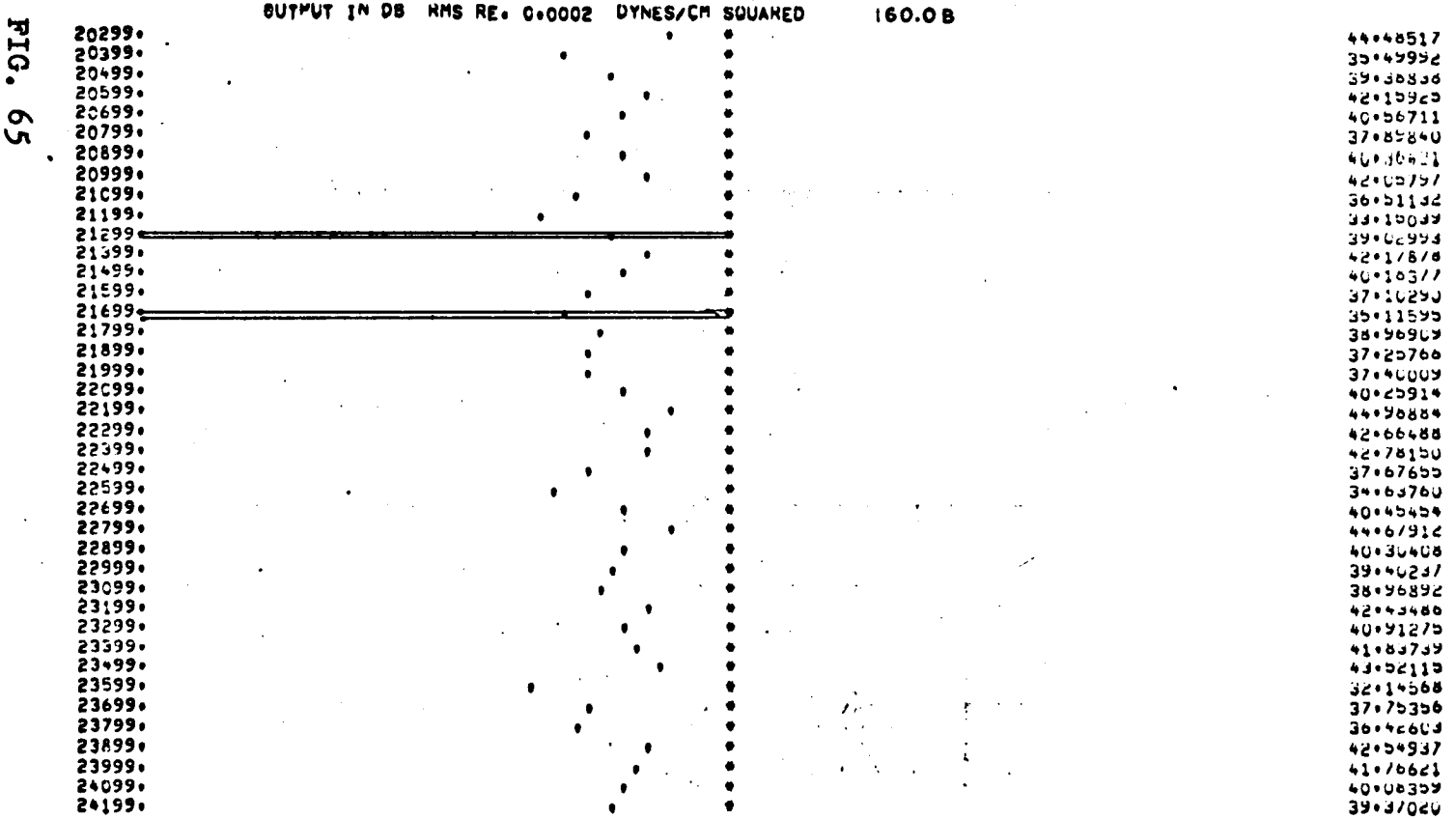


FIG. 65

© UNIVAC DATA CORPORATION NEW YORK

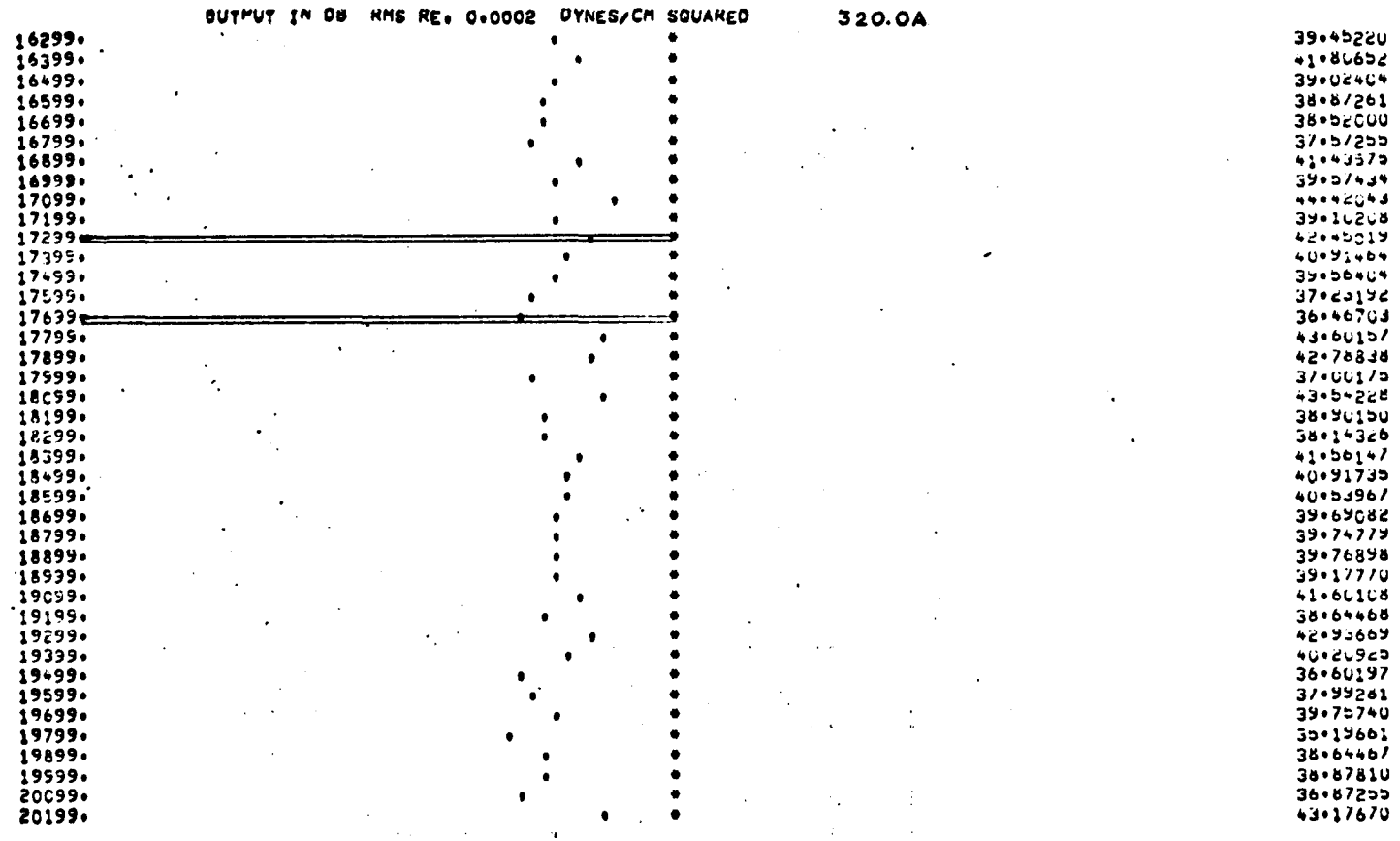
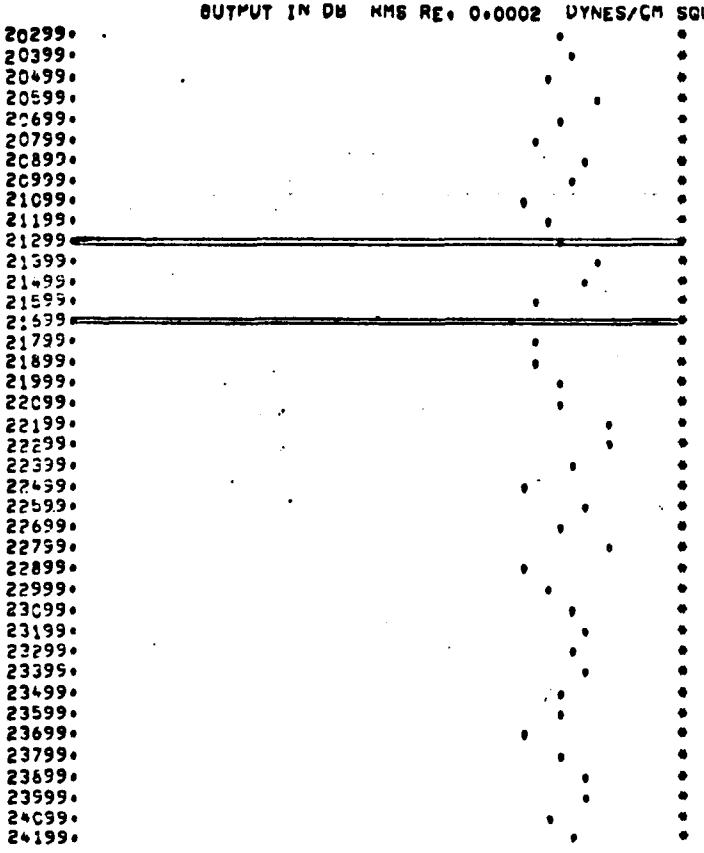

OUTPUT IN DB RMS RE. 0.0002 DYNES/CM SQUARED
320.0A
16299. 39.45220
16399. 41.80652
16499. 39.02404
16599. 38.87261
16699. 38.52000
16799. 37.57255
16899. 41.49575
16999. 39.57434
17099. 44.42043
17199. 39.10208
17299. 42.45019
17399. 40.91464
17499. 39.56404
17599. 37.29196
17699. 36.46703
17799. 43.60157
17899. 42.78838
17999. 37.00175
18099. 43.54228
18199. 38.90150
18299. 38.14326
18399. 41.56147
18499. 40.91735
18599. 40.53967
18699. 39.69082
18799. 39.74779
18899. 39.76898
18999. 39.17770
19099. 41.60108
19199. 38.64468
19299. 42.93669
19399. 40.20925
19499. 36.60197
19599. 37.99261
19699. 39.75740
19799. 35.19661
19899. 38.64467
19999. 38.87810
20099. 36.87255
20199. 43.17670
MEAN-DB= .100C000002E+01 STAND. DEV. OF MEAN= .2564113721E+00 NORM. DEV. = .2564113715E+00
TRIAL NUMBER C HAS BEEN TAPED

FIG. 66

OUTPUT IN DB RMS RE. 0.0002 DYNES/CM SQUARED 320.8



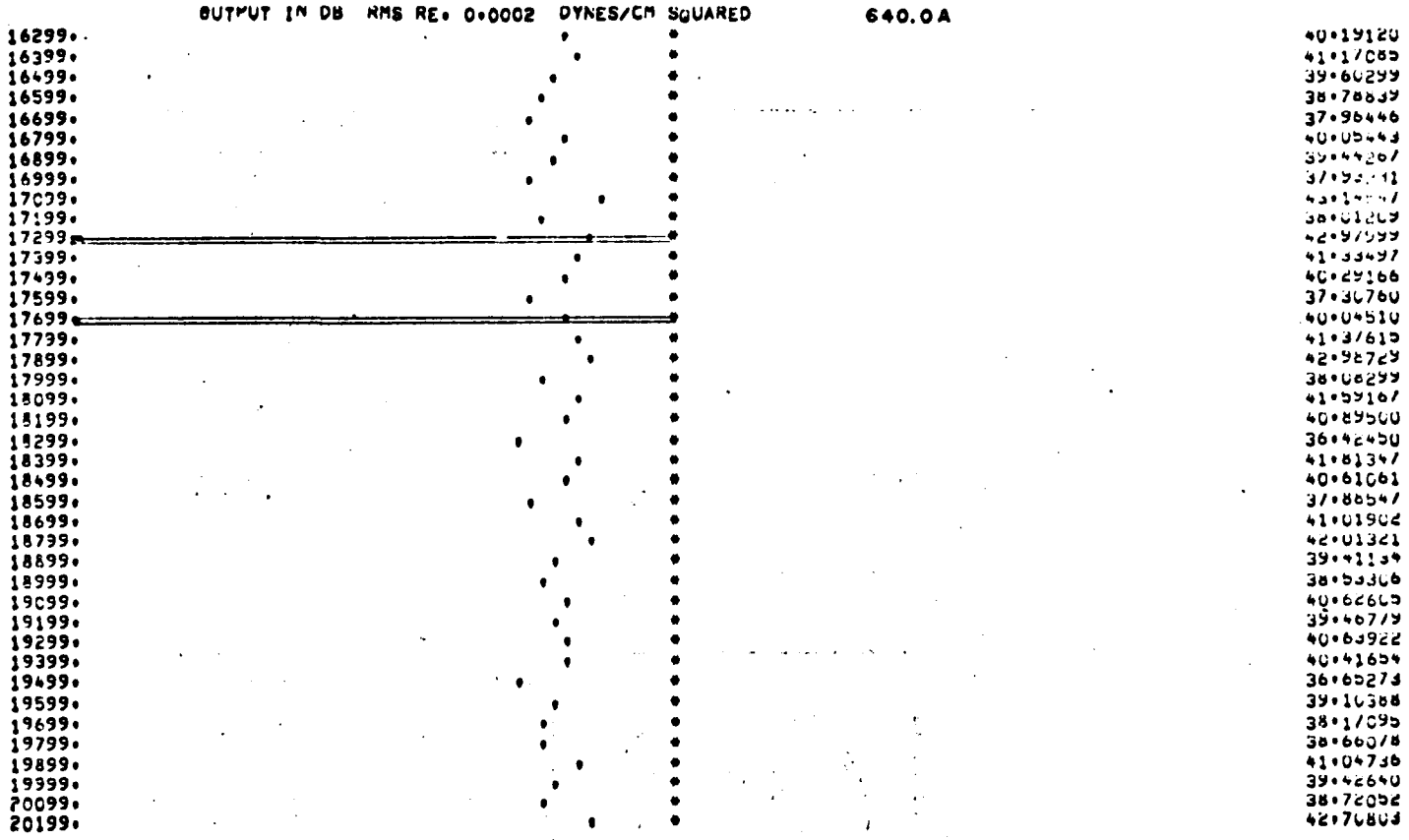
20299.	39.70902
20399.	40.21305
20499.	38.88753
20599.	42.72453
20699.	39.97804
20799.	37.56460
20899.	41.01110
20999.	40.14248
21099.	36.9[illegible]74
21199.	38.44237
21299.	39.01811
21399.	42.26256
21499.	41.41591
21599.	37.27443
21699.	35.19523
21799.	37.84053
21899.	37.01640
21999.	39.91690
22099.	39.56210
22199.	43.77527
22299.	43.02157
22399.	40.77775
22499.	36.94559
22599.	41.47746
22699.	39.20347
22799.	43.47564
22899.	36.48986
22999.	38.99060
23099.	40.91830
23199.	41.83698
23299.	40.66765
23399.	41.18832
23499.	39.89055
23599.	39.63383
23699.	36.79001
23799.	39.21624
23899.	41.18148
23999.	41.70680
24099.	38.05950
24199.	40.35518

MEAN.DB. .9999999996E+00 STAND. DEV. OF MEAN. .2375205149E+00 NORM. DEV. . .2375205150E+00

TRIAL NUMBER C HAS BEEN TAPED

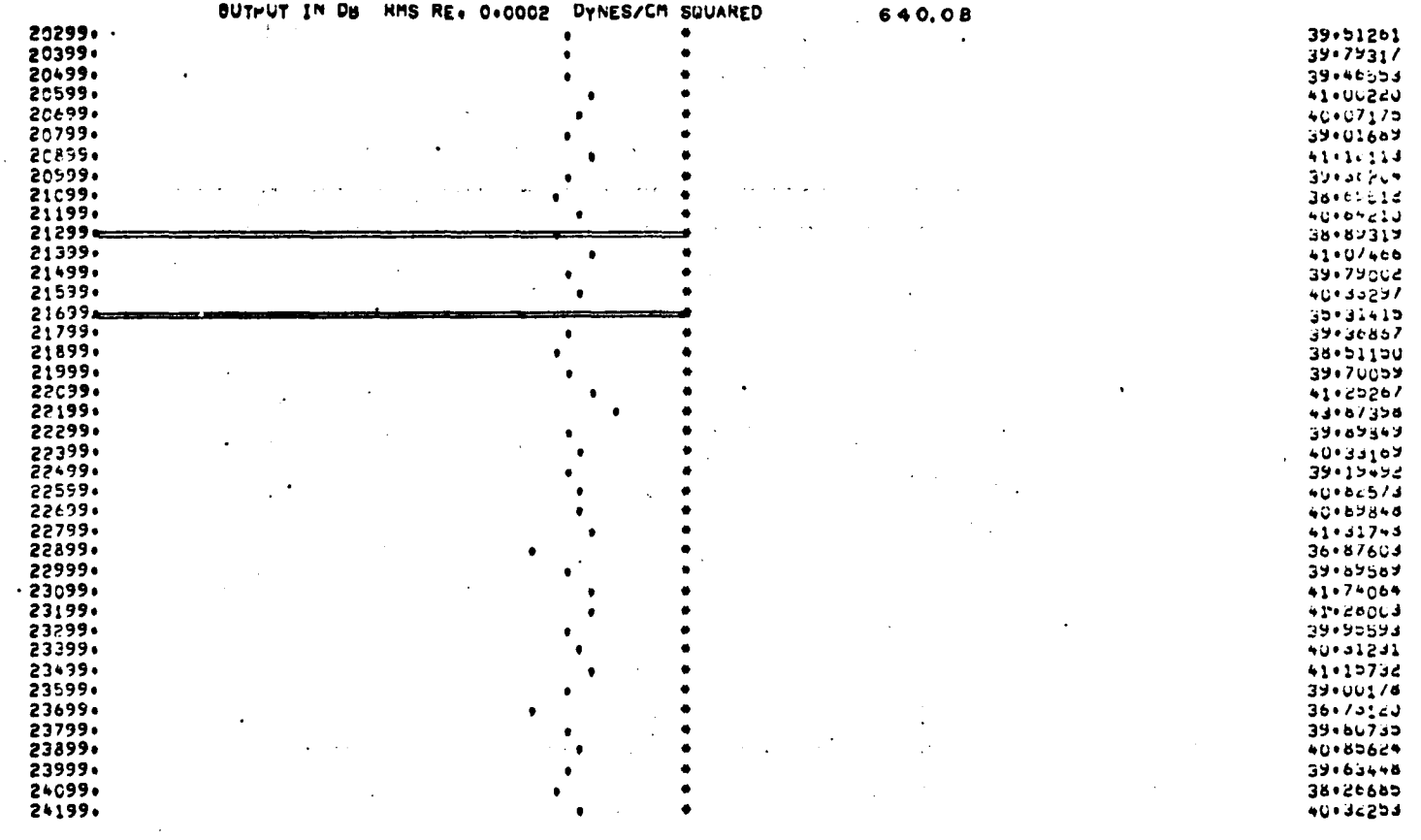
FIG. 67

FIG. 68



© NATIONAL DATA CORPORATION

FIG. 69



MEAN-DB= .1000000004E+01 STAND. DEV. OF MEAN= .1670038157E+00 NORM. DEV. = .1670038101E+00

TRIAL NUMBER C HAS BEEN TAPED
EXIT

The appropriate trends were found only for noise segments approximately 40 msec. long, occurring in the period coincident with, or almost coincident with, the signal onset period. (Fig. 49).

In addition to a similarity in trend between these curves and the resulting thresholds, a quantitative correspondence also exists at the 2.5 and 20 Hz bandwidths where it will be shown Factor 2 effects do not appear to exist.

At the 2.5 Hz bandwidth, both noises are relatively flat exhibiting no signal-like envelope excursions (Figs. 52 and 53). However, the LIM is about 5 dB greater for Noise B than for Noise A. These differences appear in the resulting thresholds, Noise B thresholds being worse for all subjects and averaging about 5.5 dB difference (Tables 11 and 12).

At the 20 Hz noise bandwidth, the LIM difference between Noise A and B is approximately 7 dB with Noise A the larger. The resulting threshold differences for the two noises for subjects 1, 2, 3, and 5 (subject 4 is omitted because of an outlier value) is 5.7, 2, 5.3, and 7.6 dB respectively in the same direction as the LIM difference, and averages 5 dB (Tables 11 and 12).

Factor 2.--Evidence of the effect upon thresholds of signal-like variations of the noise envelope as predicted by the hypothesis, is found at noise bandwidths of 5 Hz and, by comparison of the thresholds generated by Noise A at bandwidths of 10 and 20 Hz.

For noise bandwidths of 5 Hz, the LIM is 9 dB higher for Noise B than for Noise A, 42 dB versus 33 dB (Fig. 49). Despite this

difference, however, Noise B produced more sensitive thresholds than did Noise A, Subjects 1, 4, and 5 showing an average improvement of 4 dB (Subject 2 gave a poorer threshold, 4 dB down for Noise B, this matter to be discussed later, and Subject 4 gave an outlier value for Noise B.) (Tables 11 and 12). This apparent inversion of results is explainable in terms of Factor 2, as follows.

Within the signal onset region, Noise A rises about 3 dB while Noise B remains relatively flat (Figs. 54 and 55). Though the Noise A envelope continues to rise after this period, its behavior within this period could result from the addition of a tone to the noise and falls within the definition of a signal-like variation as defined by the hypothesis. This behavior degrades the subjects detection sensitivity for Noise A and produces the observed results.

A comparison of the thresholds arising from Noise A at bandwidths of 10 and 20 Hz with the LIM difference, also reveals the effects of Factor 2. Though the maximum LIM deviation from the Noise B line, occurs at 20 Hz, the derived thresholds reach their maximum deviation at a noise bandwidth of 10 Hz for all subjects not having outlier values (ie Subjects 1, 2, 3, and 5 Figs. 49 and Tables 11 and 12). Figures 56 and 58 reveal the presence of both factors, an elevated LIM and a signal-like noise envelope variation for the 10 Hz bandwidth noise within the signal onset period. Though the 20 Hz bandwidth noise also contains envelope excursions in this region they are not signal-like, rising much faster than the signal, ie. in 20 msec, and falling off as rapidly afterwards. The result of both factors acting

in concert at the 10 Hz noise bandwidth, deteriorates these thresholds and shifts the peak of the threshold curve downward to this bandwidth.

Wider Bandwidths.--For wider bandwidths the effects of both factors diminish. The sampling interval encompasses more independent samples, as bandwidths increase, and as a result, the variance of the means diminishes. This tendency can be seen for the LIM at the 80 Hz noise bandwidth in Figure 49 and also in Figures 50 to 69 which show noise waveforms averaged over 10 msec. intervals. The effects of Factor 2 are also diminished for widening bandwidths, since the envelope excursions become more rapid and less signal-like. For supra-critical bandwidths, subjects reported the entrance of another factor, a change in quality of the presentation upon the addition of the signal. However the main investigation supplies no data on this matter beyond this subjective report.

Subject 2.--The reaction of Subject 2, can be tentatively explained. Subject 2 was the only naive listener of the five and reported using only a general overall impression of comparative loudness as the basis for decision. Despite this, however, the subject's threshold for Noise A is only slightly poorer at the 1.25 Hz noise bandwidth than that for all other subjects while her other Noise A thresholds are similar to those of the remaining four subjects. The subject's Noise B thresholds do differ, however, from all other subjects, in that they cannot be fitted by a straight line. In addition, the thresholds do not show the reversal of sensitivity as compared to local noise mean intensity, shown

by all other subjects for the 5 Hz noises.

Though the subject reported responding only to overall presentation loudness differences, this should produce straight, not curved threshold sections for both noises nor should the threshold differences between the effects of the two noises occur. A parsimonious explanation suggests that the subject did not listen as intently as the others for the change in the presentation envelope denoting signal presence and, as a result, responded more strongly to Factor 1 than to Factor 2. When the noise envelopes did change widely, as in the 10 and 20 Hz cases, the subject reacted by raising her criteria regardless of the direction of change. Support for this explanation may be garnered by noting that upon making allowances for the elevation of the 10 and 20 Hz thresholds, the subject's threshold curves follow closely the Factor 1 intensity curves of Figure 49 (or, conversely, displacing the 10 and 20 Hz intensities upward, causes the curves to mirror the subject's thresholds).

Results Arising From Outlier Values

An Hypothesis Concerning Outliers.--Up to this point, four extremely low thresholds, established as outliers, have been excluded from the discussion. These thresholds occur at -14 dB for the 5 Hz bandwidth Noise A for Subject 1, -16 dB for the 5 Hz bandwidth Noise B for Subject 2, and at -17.8 dB and 17.6 dB for the 10 and 20 Hz bandwidths of Noise B for Subject 4 (Table 15).

There are three reasons why these results cannot be dismissed as merely accidental: (1) these thresholds are unusually sensitive

compared to both the results reported in the literature and those arising from the non-outliers in this study, (2) the corresponding psychometric curves are significantly flatter than the non-outlier curves indicating that a difference in detection mode may exist, and (3) the subjects giving these outliers expressed the conviction that they had caught on to and retained the pattern of the noise. The hypothesis that a unique mode of detection, involving some form of pattern discrimination, suggests itself.

Support for the Hypothesis

Frequency Discrimination.--Some evidence that another more sensitive mode of perception exists than that functioning in the typical detection test, is found in the data arising from studies of frequency discrimination.

The discrimination of changes in frequency is a task that is recognized as differing in kind from either intensity discrimination or the usual detection task. Steven's (1934) noted this difference between sensations varying in quality and those varying in magnitude and observed that they obeyed different laws of growth. Wightman (1973) supported this distinction and proposed that pitch perception is a pattern recognition process.

Shower and Biddulph (1931) measured the difference limen for frequency. Their results when transformed to the same form used to express the results of detection tests, ie. the ratio of the frequency change increment to the base frequency, give results as low as -27 dB at 800 Hz. This result indicates that the discrimination of quality

changes, a process possibly related to pattern discrimination, can be more sensitive than detection based on changes in magnitude and lends support to the hypothesis that the outlier thresholds may be the result of a related process involving quality or pattern distinction.

Psychometric Slopes.--Green (1960) noted that the application of signal detection theory to signal detection in noise, predicts a decreasing steepness (smaller slope) of the psychometric curve, in the region surrounding threshold, as the degrees of freedom (number of uncertanties) decrease. The strong expressions of confidence about their test results displayed by subjects for the tests giving outlier thresholds is reflected by the smaller (flatter) slopes of their psychometric curves in accordance with Green's discussion. Table 16 displays the slopes and the thresholds for all noise bandwidths giving outlier thresholds. The flatter slopes, below .159, appear, without exception for all outlier threshold values. However, two non-outlier thresholds, Subject 5 at 10 Hz for Noise A and Subject 1 at 5 Hz for Noise B also gave flatter slopes.

V. Comparison to Other Studies

A. General

In this section the results, hypotheses, and conclusions arising from the main investigation are compared with those of previous investigators. The discussions which follow refer to Figures 70 through 74 which display the results as the ratio of the signal intensity at

TABLE 16

Slopes of Psychometric Curves Containing Outlier Values

Subj.	Threshold	Noise Band	Slope
1	-13.98	10A	0.158
2	+05.93	10A	0.522
3	+03.37	10A	0.645
4	-17.83	10A	0.134
5	+05.05	10A	1.28
1	-00.57	20A	0.773
2	+03.11	20A	0.773
3	+02.31	20A	0.586
4	-17.63	20A	0.145
5	+02.82	20A	0.821
1	-07.32	5B	0.105
2	+00.57	5B	0.465
3	-16.62	5B	0.155
4	-08.78	5B	0.405
5	-06.30	5B	0.206

FIGURE 70: Thresholds versus Noise Bandwidths found by Investigators Discussed in Text

The abscissa gives the noise bandwidth; the ordinate, the signal intensity to noise spectrum level at threshold. The following legend applies to this Figure and to Figures 71 through 74 as well.

" Δ "	Fletcher (1940)
" • "	Schafer, Gales, Shewmaker, and Thompson (1950)
" O "	Hamilton (1957) tone duration--400 msec.
" o "	Hamilton (1957) tone duration--200 msec.
"□ "	den Brink (1954) tone duration--500-1000 msec.
"⊠ "	den Brink (1964) tone duration--200 msec.
"▲ "	Bos and de Boer (1966)
" X "	results of present study Noise A
" ▽ "	results of present study Noise B

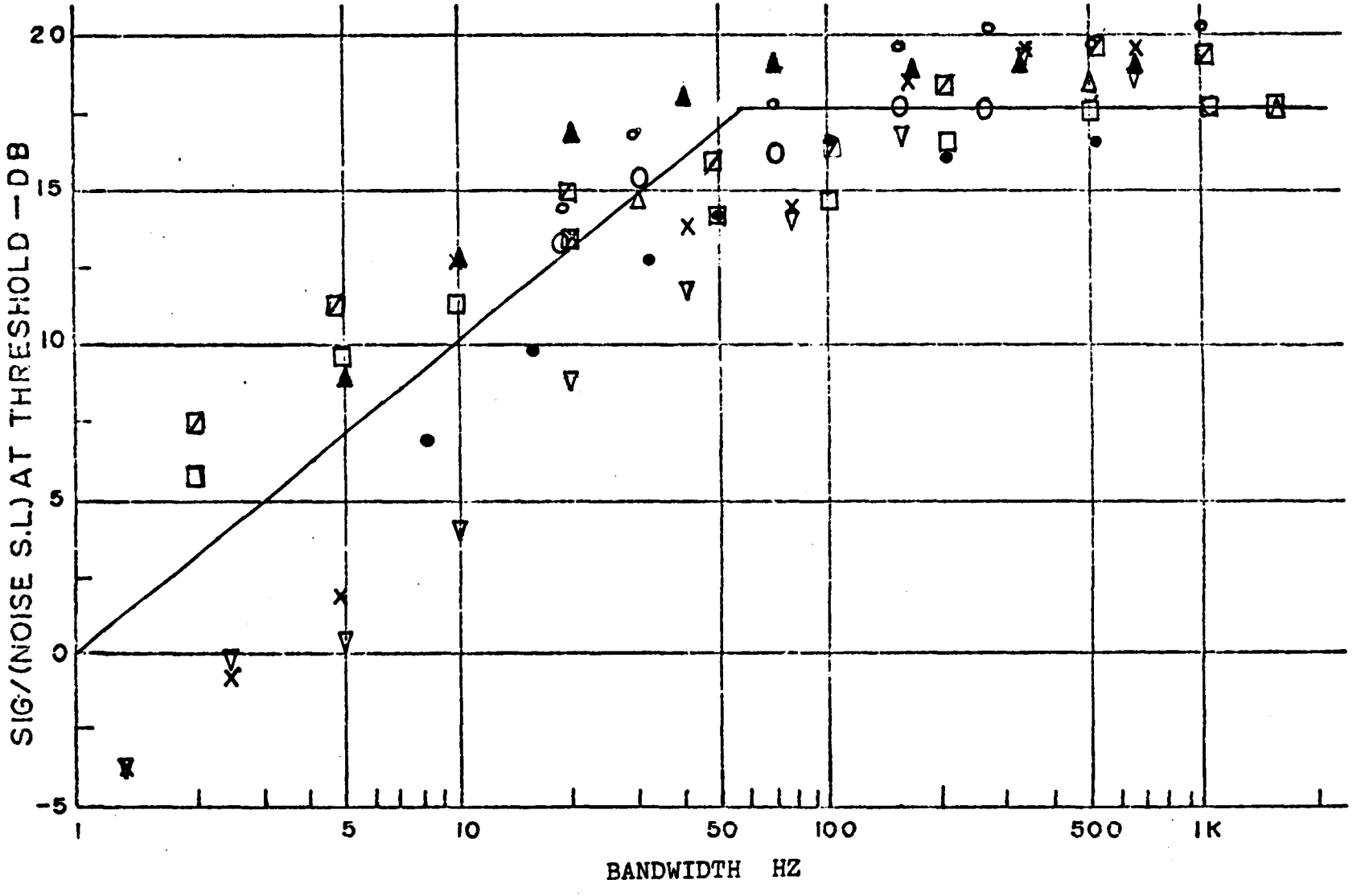

SIG/(NOISE S.L) AT THRESHOLD — DB
20
15
10
5
0
-5
1
5
10
50
100
500
1K
BANDWIDTH HZ

FIGURE 71: Threshold versus Noise Bandwidth Curves Found by Fletcher, Schafer et al, and Present Study

Abscissa and ordinate as in Fig. 70. The legend is as follows:

"___" Fletcher (1940)

" • " Schafer et al (1950)

" X " present study Noise A

" ∇ " present study Noise B

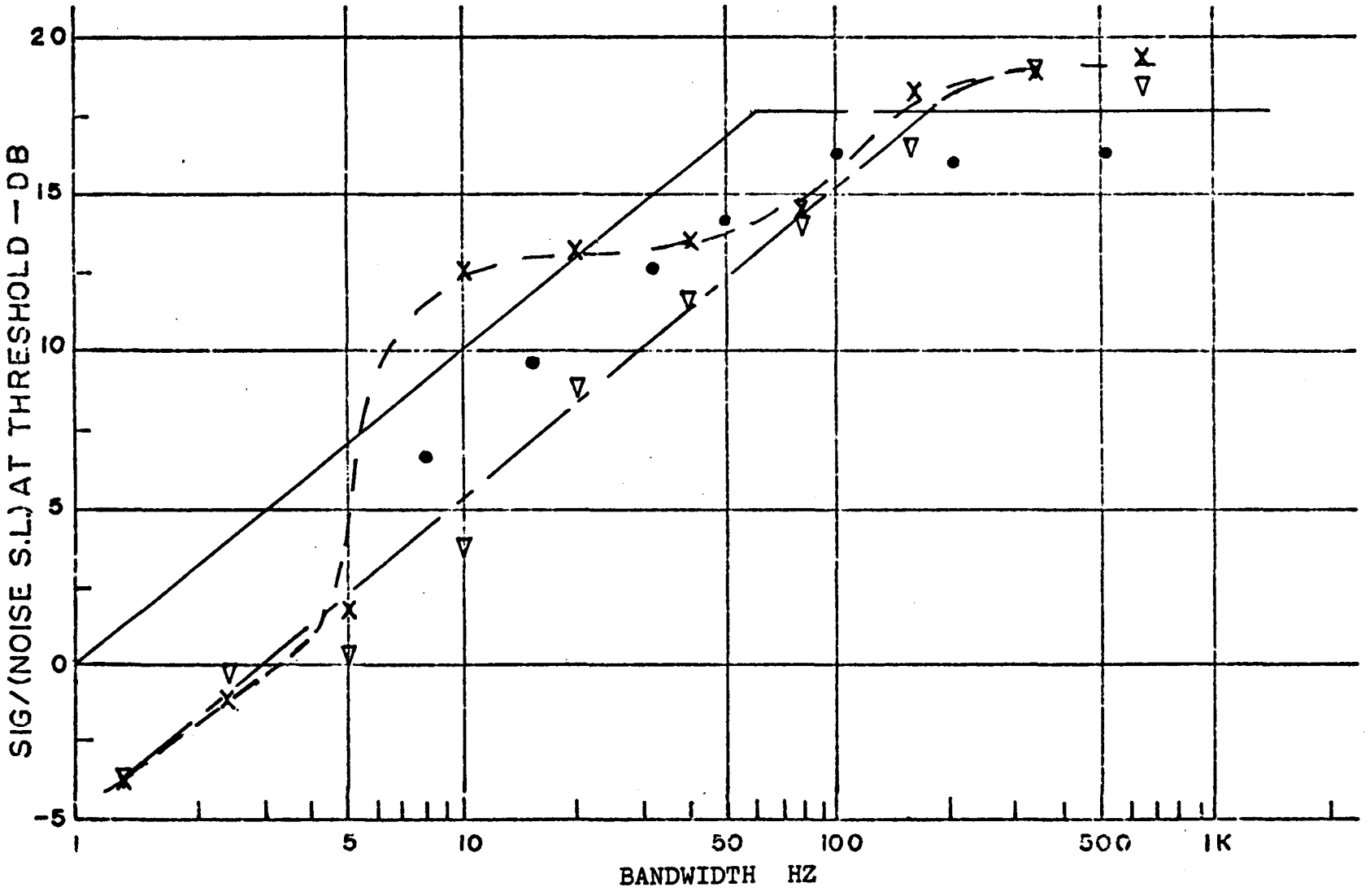

SIG/(NOISE S.L.) AT THRESHOLD — DB
20
15
10
5
0
-5
1
5
10
50
100
500
1K
BANDWIDTH HZ

FIGURE 72: Threshold versus Noise Bandwidth Curves Found by Fletcher, Hamilton, den Brink, and Present Study

Abscissa and ordinate as in Fig. 70. The legend is as follows:

"___" Fletcher (1940)

" o " Hamilton (1957) 200 msec.

" ☒ " den Brink (1954) 200 msec.

" X " present study Noise A

" ▽ " present study Noise B

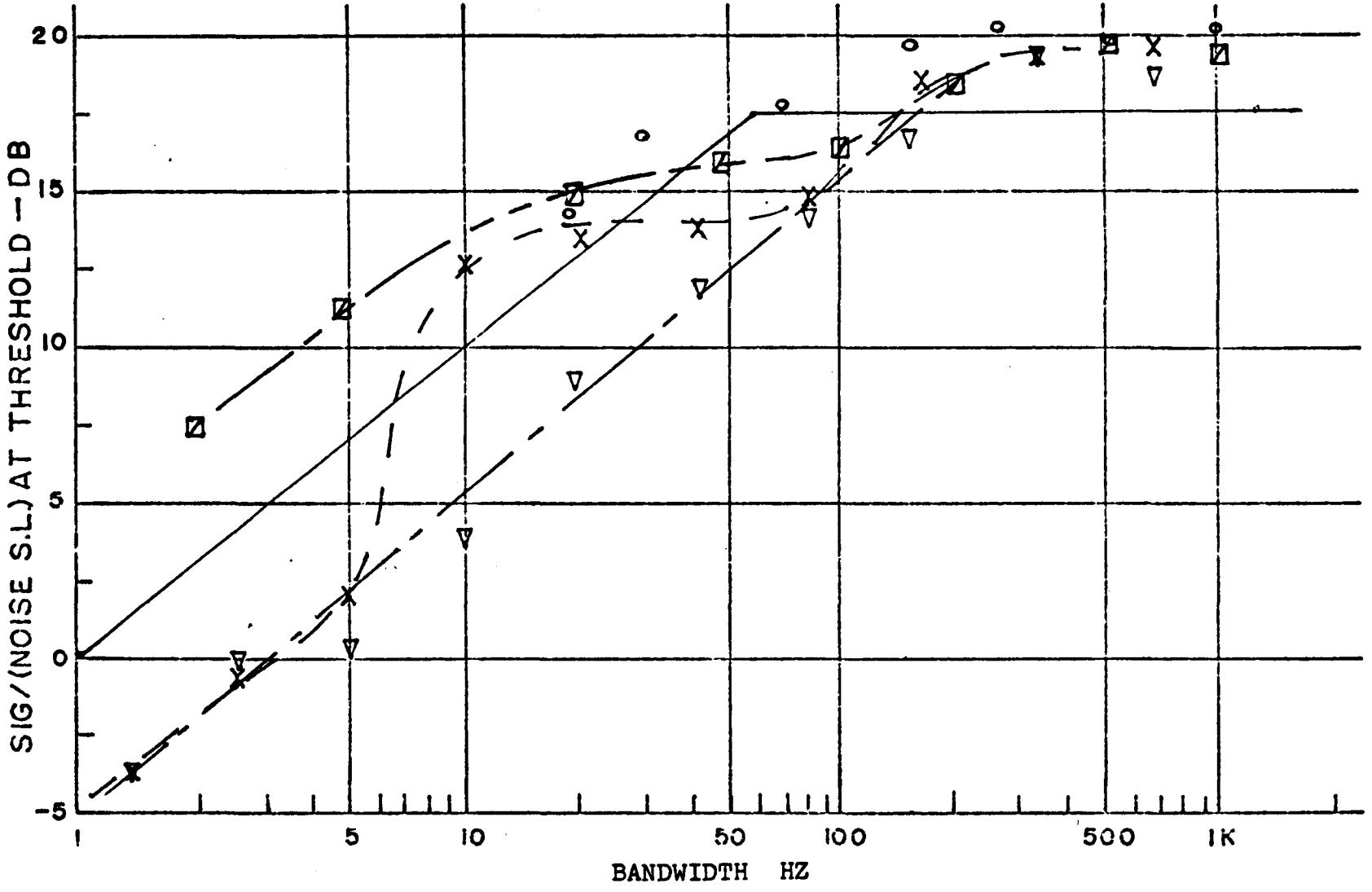

SIG/(NOISE S.L.) AT THRESHOLD — DB
20
15
10
5
0
-5
1
5
10
50
100
500
1K
BANDWIDTH HZ

FIGURE 73: Threshold versus Noise Bandwidth Curves Found by Fletcher, Hamilton, Bos and de Boer, and Present Study

Abscissa and ordinate as in Fig. 70. The legend is as follows:

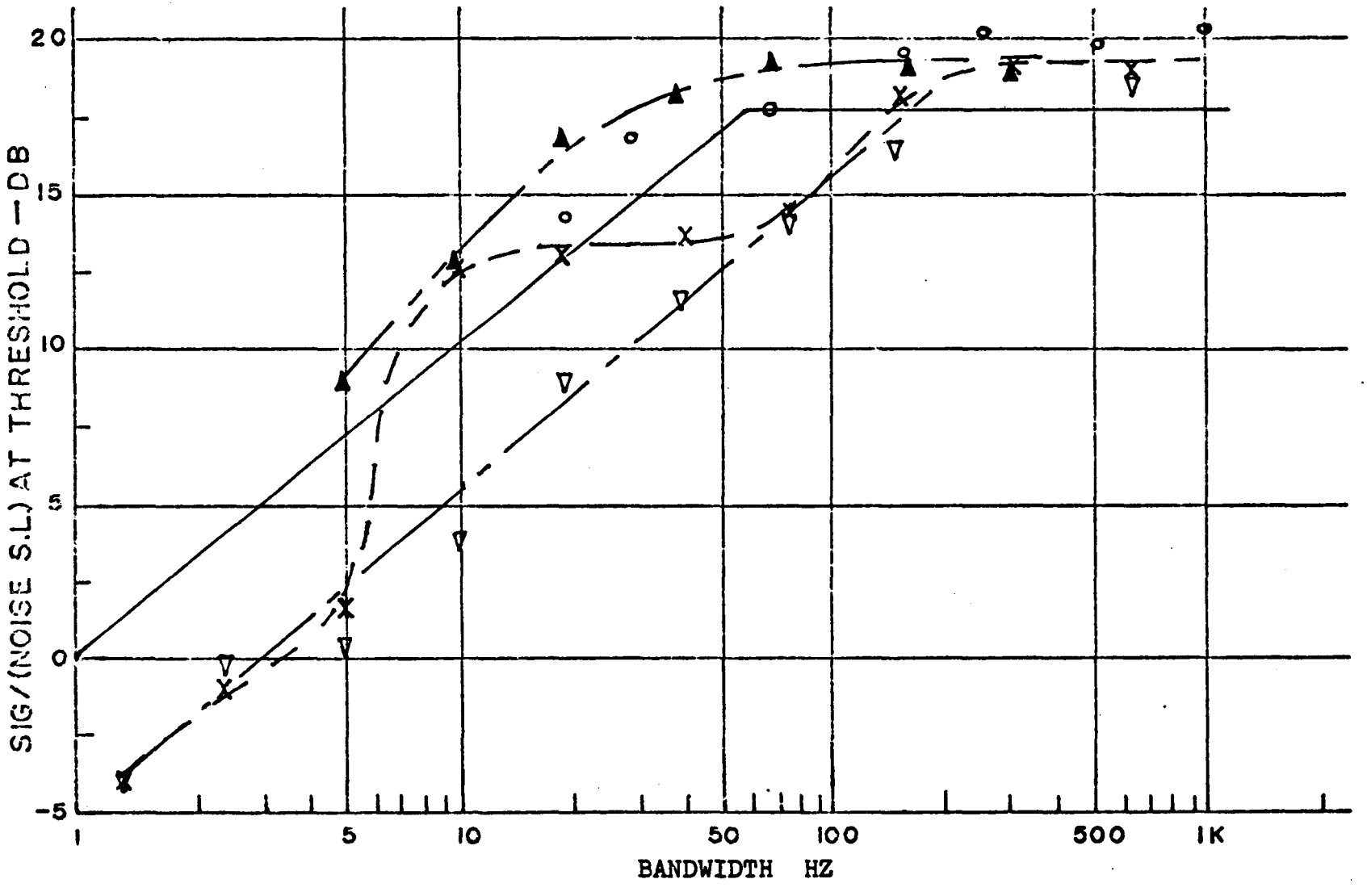
"___" Fletcher (1940)

" o " Hamilton (1957) 200 msec.

" ▲ " Bos and de Boer (1966)

" X " present study Noise A

" ▽ " present study noise B


SIG/(NOISE S.L) AT THRESHOLD — DB
20
15
10
5
0
-5
1
5
10
50
100
500
1K
BANDWIDTH HZ

threshold to the noise spectrum level versus the noise bandwidth. Figure 70 displays the derived results for all investigators under consideration, while Figures 71 through 73 display the approximate results of the main investigation with groupings of results drawn from Figure 70 so as to add clarity to the discussion. In these figures, a single approximate curve has been drawn to represent the results of the main investigation for Noise A thresholds and similarly for Noise B. This is done for convenience of comparison only and no suggestion is intended that these curves represent anything but approximate trends. Because of variations in the techniques employed by the several investigators, the following discussion focuses more on a trend and slopes of curves than on absolute magnitudes.

B. Differences Between Fletcher and den Brink

From the viewpoint of trends, the set of curves indicates that the data of Fletcher (1940), Schafer et al (1950), Bos and de Boer (1966), and the Noise B curves of the main investigation are best fitted, sub-critically, by a straight line with a slope of 3 dB per octave of noise bandwidth, while the data of Hamilton (1957), den Brink (1964) and the Noise A curves of the main investigation require curved lines (note Hamilton's data is considered to be fitted by den Brink's curves as is asserted by den Brink (1964). See Figure 72.

The most striking difference in sub-critical threshold curves arising from previous investigations, is exemplified by Schafer et al on the one hand and den Brink on the other (Figure 70). There is, as well, an important difference in signal-generation techniques between

the two, as regards their treatment of the pure tone envelope, which merits scrutiny.

Schafer et al did not control the shapes of the initial and final transitions of the pure tone signal relying apparently upon the frequency limitations of their system to eliminate clicks. Since their system accommodated noises with center frequencies as high as 3200 Hz and bandwidths of 1000 Hz it is evident that signal transitions could be very fast when compared to the rate at which the envelope of sub-critical noises could change. As noted earlier, this reflects a disregard of the effects of spectral dispersion of the signal.

On the other hand, den Brink and Hamilton were concerned about these effects and, consequently, processed the pure tone pulses through the same filters used for the noise bandwidth under test. This procedure limits the spectral dispersion of the signal, but it reflects in turn, a disregard for the effects upon thresholds of envelope transitions and other resulting distortions.

This disparity in signal processing techniques between the two representative investigators is suspect in light of the disparity of their results. The results of the main investigation provide a means of possibly explaining and reconciling this basis difference.

Reasons as per Hypothesis

For presentations of fixed mean intensity, two additional factors were held to influence the course of the sub-critical threshold curves by the main investigation. These were local variations in mean intensity in the region of signal onset and, in the same region, changes

in the noise envelope at rates similar to signal transitions. These two factors, though related by the duration of the signal transition may exist and exert their effects independently. It was noted, as well, that the existence of a specific noise bandwidth region in which the deviation of the local intensity mean of the noise envelope from the presentation mean could be expected to be maximized, was also a result of the special conditions of the experiment. It will be shown that other conditions exist for the studies under discussion and that, under these other conditions, Factor 1 is minimized while Factor 2 remains significant.

Consider again the restraints upon envelope changes held to exist as a result of the conditions of the main investigation. The maximum Factor 1 envelope deviation from the presentation mean was limited for very narrow bands by the maximum rate permitted by the particular noise bandwidth and by the duration of the noise. For wider bands, the Factor 1 intensity deviation was a function of the number of independent samples encompassed within the signal transition period.

Since Schafer et al, Hamilton, and den Brink, both used much longer noise durations than those used in the main investigation, 5 sec., 2 sec. and continuous noise, respectively, this first effect, the limitation on Factor 1 deviation by the noise passband is either removed to bandwidths narrower than those used by any of the investigators, or eliminated entirely as is the case of continuous noise. In addition the absence of control over the presentation mean minimizes this effect even for noises of short duration.

The second limitation on Factor 1, the number of independent samples encompassed within the signal transition period remains in effect but this, it should be noted, is a function of the signal transition period. This is also true for Factor 2, signal-like changes in the noise envelope, which also remains operative.

The signal transitions used by Schafer et al, though unspecified, are probably so abrupt that their rate of change exceeds greatly the rate of change of the sub-critical noise envelope. This means that because of this very short sampling period the number of independent samples occuring within the signal transition period is never more than one for all sub-critical noises. Effectively then, the tendency for maximum Factor 1 deviations to be confined to any restricted frequency range is eliminated in the experiments of Schafer et al since both conditions for the presence of this factor, relatively short noise pulses and relatively significant transition periods, are absent.

These same abrupt signal transitions also minimize, or eliminate, the operation of Factor 2 since there can be no changes in the envelope of the sub-critical noise fast enough to mimic the abrupt change attendant upon the introduction of the signal.

The theoretical discussion presented above can be translated to simpler language. It says, in effect, that the abrupt onset of the signal minimizes the chances that this transition will coincide with local intensity means which deviate more in one region of the sub-critical bandwidth than in any other. It also says that this same abrupt signal transition minimizes the chance that the slower changes of the

noise envelope will be mistaken for it.

The elimination of both factors as significant agents in the experiments of Schafer et al, leaves only the mean signal to noise intensity ratio (or other unknown factors) as remaining factors. A sub-critical line with a slope of 3 dB per octave, as found by Schafer et al, is the expected result. See Figure 71.

The signal transitions used by den Brink and Hamilton, are neither fixed nor abrupt. Instead they are functions of the bandwidth of the noise which they accompany since both signals and noise are processed through the same filter. For this reason, moreover, the noise envelope is always likely to change at the signal transition rate as soon as the noise filter begins to restrict the signal transition rate. This condition leads to noise presentations which may be rich in signal-like envelope changes for most of the sub-critical noise bandwidths. Under these conditions, a significant degradation of threshold sensitivity would be expected to occur at noise bandwidths in which changes in level constitute the major detection clue. Presumably this would begin to be true for noise bandwidths just within the critical band borders where the noise begins to sound like a fluctuating tone, and continue to the narrowest bandwidth. Den Brink's curve reflects this action. (Figure 61).

An additional element present in den Brink's procedure, the random initiation of his signals in continuous noise, could also increase the effect of Factor 2 by causing the subject to remain continuously primed to respond to signal-like variations of the noise envelope throughout

the test.

The suggestion that fluctuations of the noise envelope degrade the sub-critical threshold, was made explicitly by den Brink (1964) who called it a criteria effect and also, more generally, by de Boer (1962) and Bos and de Boer (1966). However den Brink did not attempt to relate this effect to signal onset time.

Den Brink also ascribed the threshold deterioration to the nature of the task changing from detection to an intensive difference limen at the narrower bands. However, since the Noise A and B threshold curves both indicate that at the narrowest bandwidths where discrimination is the likeliest process, threshold sensitivity returns to its initial value, prior to the bulge, his explanation is not supported by the main investigation. The supposition, discussed above, that the confusion engendered by Factor 2 causing both increased false alarms and a raising of the subject's criteria provides an explanation of den Brink's results more harmonious to the main investigation.

The disparity in sub-critical threshold curves found by Schafer et al to those of den Brink is apparently due to differences in technique which inhibit or stimulate the operation of at least two factors which have the potential to influence sub-critical thresholds. In the investigation of Schafer et al, the experimental conditions appeared to inhibit both of these factors leaving average signal to noise intensity ratios as the dominating influence. Den Brink's conditions, on the other hand, apparently permitted the confusions caused by signal-like changes in the noise envelope to degrade detection sensitivity over most of the sub-critical band. The influence of testing conditions and

fortuitous noise envelopes upon the operation of factors, other than long term signal to noise ratios, is further illustrated by the two curves of the main investigation. Curve A illustrates the results of the two factors, 1 and 2, in apparent summation while curve B shows the results when they are both absent. While Curve B is similar to that of Schafer et al, Curve A presents an entirely new result. The suspicion is aroused that testing under other conditions, or with other noise samples, would produce more new results as well.

C. Other Results

The hypothesis, suggested above, is supported by the threshold curve of Bos and de Boer (1966). This curve (Figure 73) though generated by pure tone signals similar to the main investigation, 250 msec long with 40 msec transition times, has a slope somewhat steeper than Fletcher's or Schafer et al's, and a critical bandwidth roughly equal to Fletcher's but narrower than Schafer et al's, Hamilton's, den Brink's, or the results of the main investigation. It shows no indications of sub-critical threshold deterioration evident in den Brink's results and those of the Noise A thresholds, the curves being closer in trend to the Noise B thresholds.

Without knowledge of the specific envelope configurations of Bos and de Boer's noise, it is not possible to determine which specific stimulus features are responsible for his results. However these results do illustrate the variety of possibilities which may occur when only the noise bandwidth is controlled and the effects of signal processing upon noise envelopes disregarded. This point is particularly

pertinent in this case because the noise signals have been subjected to unusual processing, single side band generation, without sufficient regard for the effects of this processing, upon the noise envelope. The effects of single side band processing have been discussed earlier but it should be recalled that it generated noise with envelopes differing substantially from those of other investigators.

VI. Conclusions re: the Main Investigation

The main investigation has suggested two factors which may affect the detection of tones in noise, the local intensity mean of the noise during the signal onset period and the presence of signal-like changes in the noise envelope during the same period. These two factors appear to be capable of causing significant deviations from a base-line threshold established by the presentation-wide mean intensity. Their effects are revealed, more clearly, by this study because the uncontrolled variations typical of many previous studies, have been eliminated by the generation technique which permits precise control of, and minute examination of the generated signals. Considerations of the findings or other investigators, moreover, indicate that these factors may produce significant effects in tests in which the presentation-wide means are uncontrolled. These effects suggest an explanation of the apparently discrepant findings of other investigators concerning subcritical detection. In addition to several abnormally sensitive thresholds indicate that another mode of detection related to quality distinction or pattern discrimination may be used by subjects when the pattern of the noise is retained in memory.

These results emphasize the dependence of the sub-critical thresholds upon the configuration of the noise envelope. They carry implications for the study of hearing which merit consideration since they derive from the realization that control of intensity alone provides only a gross tool which is unable to resolve important sub-critical detection mechanisms and that such resolution requires control of, and knowledge about, the noise envelopes as well.

Specifying the filter configuration, even when attention is paid to magnitude, phase, or transient characteristics, does not determine all the important features of the noise stimulus. Filter characteristics only establish upper bounds upon the rate at which the noise envelope may change. The specific characteristics of the noise at the output of a filter also depends upon its characteristics at input. The probability always exists that a specific noise presentation, after filtering, will have an envelope configuration more appropriate to a filter narrower than that used. Moreover, it is clear that the number of possible envelope shapes of noise with a fixed maximum rate of envelope change is immense and cannot be controlled by specifying a few derivative characteristics. When attention is paid only to the bandwidth of the filter, as is typical, the envelope of noise processed by such a filter is not fully specified.

Since sub-critical thresholds appear to be so strongly influenced by the noise envelope, a complex configuration of potentially great variety, the existence of many yet unconsidered factors is a possibility. The hypothesis advanced to explain the unusually sensitive outlier

thresholds suggests that some of these factors may be highly sophisticated utilizing higher order functions of the human brain, perhaps involving primitive aspects of speech detection and comprehension. This suggests, in turn, that while the determination of the effects of the more traditional factors, such as intensity, duration, and bandwidth retain their interest and importance, care is required in such investigations to prevent the intrusion of higher level factors. These potential, new factors are, in their own right, of interest since in addition to revealing new and more sophisticated detection mechanisms, they may provide a bridge between non-speech and speech. Both kinds of investigations require precise control of the noise envelope.

The considerations discussed above indicate that the tacit acceptance of global parameters such as the bandwidth of the processing filter, long term statistical averages, etc., as the only valid parameters of noise, misleads the researcher into viewing them as the only valid factors affecting hearing. The more fundamental viewpoint starts with the configuration of the noise envelope and also includes its various kinds of averages. The researcher should be able to frame his test specifications directly in these terms, adjusting presentation means, local intensity variations, envelope excursions for both repetitive and non-repetitive noises so as to test specific hypotheses. It should be emphasized that to frame such hypotheses the traditional parameters should not be abandoned, but instead, included in the wider ensemble of potential factors.

For example, the major hypothesis concerning the two factors, ad-

vanced by the main investigation, requires further testing. A definitive procedure for the testing of this hypothesis requires isolation of these factors in the noise presentations by holding one constant while the other is adjusted. In this case the independent parameter will be something other than noise bandwidth, leading to curves of threshold versus local intensity level in the region of signal transition or threshold versus noise excursions of fixed height with rate of change of excursion as a parameter.

The versatility and precision of control required for the envisaged investigation lie beyond the present capability of hardware generated noise (including digital sources). Though the required control of the noise envelope is included in the present computer generation program, only indirectly via the multiple pass technique, more efficient and more versatile means of direct envelope control lie easily within its capabilities. One such extension of capabilities is discussed in Chapter V.

CHAPTER V.

CONCLUSIONS: OVERALL

I. Introduction

The main investigation, described in Chapter IV, was undertaken both for its intrinsic interest and to evaluate the utility of the computer generation program for psychoacoustic research. In the course of supplying this evaluation, the investigation also provided suggestions for the extension of the program as to increase its effectiveness. In the following sections the generation program is evaluated on the basis of the experience garnered in the conduct of the main investigation. Subsequently, suggested extensions of this program are considered.

II. Evaluation

The major expectations for the generation program, as used in the main investigation, were that the difficulties heretofore associated with the generation of narrow band noise would be eliminated, that the precision of control over the noise would increase so that some parameters not previously controllable, the mean intensity of individual presentations and the starting phase of the pure tone signal, could be precisely specified, and that the detailed record of the noise, residing in computer-accesible memory, would facilitate analysis of the effects upon hearing, of stimulus features. It was also expected that the use of the generation program together with the battery of programs

assisting both the preparation of input data for the generation program and the analysis of the results, would bring the large scale investigations required by psychoacoustics within the capabilities of a single individual. This last expectation was considered to be of great practical importance.

All expectations were satisfied. With the normal assistance of the computer facility staff, signals running 5 hours in real time were generated and transferred onto audio tape with much less effort and time, on the part of the investigator or anyone else, than would have been required for an equivalently scaled investigation using hardware generation. The recorded signals and noise were satisfactory (a few tapes had to be redone because of equipment problems). The retention of the noise signals in computer-accessible memory did facilitate the search for stimulus features affecting detection by permitting scrutiny of the noise envelope and by facilitating the calculation and display of localized intensity means, thresholds, and psychometric curves. While the use of the computer for such analysis and display is usual practice, nowadays, this usage together with the efficiencies gained from the generation program, itself, did permit a single investigator to carry out a full scale psychoacoustic investigation. Moreover, without the degree of control over the presentation parameters, made possible by the use of the generation program, the depth and resolution of the main investigation would not have been possible.

Not all the efficiencies potential to this system were realized however. Lack of appropriate equipment prevented feeding the subject's

responses either into a digital storage device or directly into an on-line computer. If the first possibility had been realized, a significant amount of hand work would have been eliminated and along with that a reduction in the possibility of error. If the second procedure had been feasible the extensive statistical analysis of results and the display of psychometric and threshold curves could have been immediately available. This would have been beneficial permitting immediate retesting of questionable results and the elimination of unnecessary testing. The digital recording of responses and the on-line usage of a computer programmed for immediate analysis is recommended.

The benefits to be gained from the use of an on-line computer together with various modifications of the generation program are attractive. The elimination of the tape recorder removes a source of possible signal contamination. The use of an adaptive program to locate thresholds as does an audio-meter operator, would reduce the amount of needed data and reduce testing time. These improvements over the procecures of the main investigation could lead to a useful psychoacoustic tool.

III. Extensions

It has been pointed out that the results of the main investigation suggest that the contours of the noise envelope play a crucial role in masking and that these contours are not fully determined by specifying the noise bandwidth. While the need for the bandwidth specification capability in the generation program remains, for a number of reasons including the fact that some relationship exists between

categories of envelopes and bandwidths, there is a parallel need for the capability to specify other, more specific noise parameters as alternative independent variables. Though this capability had been incorporated into the original program, its importance had not been fully comprehended prior to the main investigation. For this reason two techniques for specifying a variety of stimulus features as independent variables, are considered below.

The first technique is presently supported, in essence, in the existing program though some modification is required. This technique requires the use of a multiple pass mode discussed previously. To apply this technique, the user selects a parameter specification directly supported by the program and related to the stimulus feature he seeks.

For example, if a signal-like change in the region of signal onset occuring at a local intensity mean equal to the presentation mean is desired, the related parameter is noise bandwidth. The kind of envelope changes desired, the acceptable range of deviations, and the bounds of the region would also be specified along with the related bandwidth. The computer would now generate the noise, search for acceptable presentations and store those found to conform to specifications. This would be continued and repeated with appropriate changes in the variable specifications until the desired ensemble of noises was filled. There are a number of problems here involving yield but they are routine. This technique should give noise presentations which can be specified by all the traditional parameters but which satisfy, as well, the specification for the desired stimulus feature.

The second technique provides a more versatile control over noise envelopes but may not permit the direct specification of traditional parameters such as the probability density distribution and the noise bandwidth. If future investigations indicate that the listener is aware of the statistical distribution in the same way that he is aware of the noise bandwidth, that is by its effect upon the noise envelope, then this technique may be useful.

To implement this method, the concept of noise generation by filtering white noise is abandoned. Instead the envelope configuration is specified by the user either analytically, perhaps choosing from a set, or by drawing the desired contour with complete freedom. The chosen contour is then used by the generation program to modulate a pure tone signal of specified frequency, producing a noise presentation with the specified envelope in a direct fashion.

As in the first technique, there are a number of routine problems to be solved, but the program is quite feasible. This approach requires breaking away from the preoccupation with traditional parameters of noise and is, probably, totally beyond the capability of hardware sources. The dependence of the investigator upon equipment both practically in terms of availability of test signals and theoretically in his hypothesizing about potential factors, is dramatically illustrated here.

IV. Summary

This section concludes the dissertation. A computer program designed to overcome the limitations of hardware signal and noise sources

and to provide more precise control of signal and noise parameters, was developed and applied to an investigation into the detection of tones masked by noise of controllable bandwidth.

Tones and variable bandwidth noise were used to determine the course of sub-critical thresholds as a function of noise bandwidths. The computer program permitted the mean intensity of each presentation and the starting phase of the tone re: the noise, to be held constant throughout the test. The program also permitted the test to encompass noise bandwidths ranging from 1.25 to 640 Hz, essentially covering the continum from tone on tone discrimination to detection in wide band noise.

Though the two repeatable noises use throughout the tests had identical presentation-wide mean intensities, two significantly different threshold-noise bandwidth curves emerged, one fitted sub-critically by a straight line rising 3 dB per octave noise bandwidth, the other, by a curve exhibiting a pronounced hump within the 10 to 40 Hz noise bandwidth region. These results, which contradict the hypothesis holding that energy, or intensity, ratios between the signal and the noise, alone determine sub-critical thresholds, were linked to two features of the noise; (1) the differences in the mean intensity of the noise lying under the on-transition region of the signal, and (2) a difference in the pattern of signal-like excursions of the noise envelope, in the same region. These factors suggested a hypothesis which was found to provide a resolution of apparent contradictions re: sub-critical thresholds reported by previous investigators.

The influence of the contours of the noise envelope upon thresholds,

shown by the main investigation, has prompted two extensions of the generation program to include a capability to specify specific envelope configurations for the masking noise.

The results of the main investigation demonstrate the advantages of computer generation of signals and noise over the use of hardware sources.

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