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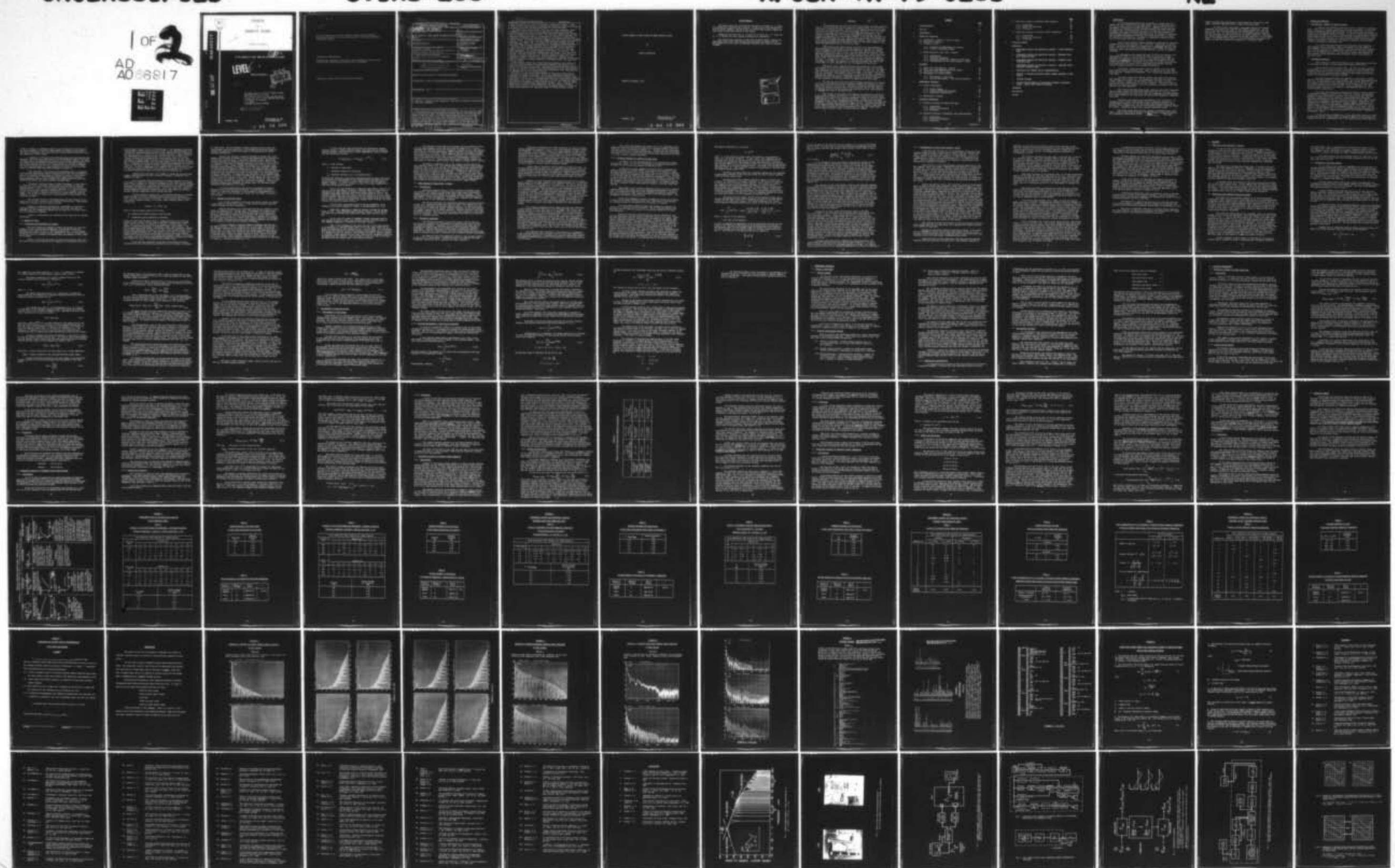

INSTITUTE FOR AEROSPACE STUDIES

UNIVERSITY OF TORONTO

ON THE LOUDNESS OF SONIC BOOMS AND OTHER IMPULSIVE SOUNDS

by

Andrzej Niedzwiecki

December, 1978

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A loudspeaker-driven simulation booth with extended rise time capability (down to 0.22 ms) has been used for subjective loudness tests of sonic-boom and other types of impulsive sounds. The first series compared N-waves over a range of 0.22 to 10 ms rise time, 100 to 250 ms duration and from 0.5 to 4 psf (the latter for the longer rise times) (24 to 192 N/m^2) peak overpressure. The response tradeoff between rise time and overpressure, and duration and overpressure was measured. For equal judged loudness, 10 ms rise time required 8 dB higher overpressure than for 1 ms. Duration had little effect in the

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SECURITY CLASSIFICATION OF THIS PAGE (When Data Entered) range 100 to 200 ms but at 250 ms noticeably enhanced the loudness. These results confirm those measured by Shepherd and Sutherland (except for the 250 ms duration), and extend the measurements down to 0.22 ms rise time.

The second series tested certain "flat top" sonic boom signatures, which according to current theory could be generated by special very long SST aircraft designed for minimized sonic-boom; these were compared for loudness with a reference N-wave (p_N = 0.5 psf, 1 ms rise time, 150 ms duration). The subjective loudness was found to be dominated by the front (or rear) shock Δp_{SH} while the maximum peak amplitude Δp_{MAX} of the inclined "flat top" had only a small effect. The results for equal loudness were well fitted by an empirical law $\Delta p_{SH} + 0.11\ \Delta p_{MAX} = \Delta p_N$. This shows that for equal loudness the peak amplitude Δp_{MAX} of the flat top signature is substantially higher than that of the N-wave; thus for equal amplitude the flat top signature is the quieter.

The third series compared filtered N-wave signatures, using a high-pass digital filter with an unfiltered N-wave signature (1 psf, 1 ms rise time, 150 ms duration). Two cut-off frequencies were used: 25 and 50 Hz. The amplitude differences for equal loudness were very slight: less than 0.6 dB at most. Thus the 'infrasonic' low frequency content of sonic boom N-waves - although it dominates the spectrum - has no significant influence on the subjective loudness. Similar tests with annoyance as a judgement criterion has showed a tendency to increase annoyance for filtered N-waves, but again this was very slight.

In the last test series the tradeoff between overpressure and duration was found for idealized quarry blast signatures composed of sequences of 25 ms long pulses with 0.22 ms rise time. The range of durations extended from 25 to 400 ms. At the short durations the loudness increased 2 dB for each doubling of duration; above 100 ms the increase was progressively lower, approaching as an asymptote the level of continuous sound.

The results in each series were compared with theoretical predictions by the method of Johnson and Robinson. All but the long-duration quarry blast judgements were found to be in very good agreement in terms of relative loudness levels. With an ad hoc - but physically plausible - modification (including adjustment of the critical integration time of the ear) the predictive method was extended to encompass the long duration signals as well. Thus the applicability of the method has been demonstrated for other types of transient sounds than the N-wave; and the extension to the method appears to bridge the range between impulsive and continuous sounds of similar spectral content.

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Abstract

1 pqm

A loudspeaker-driven simulation booth with extended rise time capability (down to 0.22 ms) has been used for subjective loudness tests of sonic-boom and other types of impulsive sounds. The first series compared N-waves over a range of 0.22 to 10 ms rise time, 100 to 250 ms duration and from 0.5 to 4 psf (the latter for the longer rise times) (24 to 192 N/m^2) peak overpressure. The response tradeoff between rise time and overpressure, and duration and overpressure was measured. For equal judged loudness, 10 ms rise time required 8 dB higher overpressure than for 1 ms. Duration had little effect in the range 100 to 200 ms but at 250 ms noticeably enhanced the loudness. These results confirm those measured by Shepherd and Sutherland (except for the 250 ms duration), and extend the measurements down to 0.22 ms rise time.

The second series tested certain "flat top" sonic boom signatures, which according to current theory could be generated by special very long SST aircraft designed for minimized sonic-boom; these were compared for loudness with a reference N-wave (p_N = 0.5 psf, 1 ms rise time, 150 ms duration). The subjective loudness was found to be dominated by the front (or rear) shock Δp_{SH} while the maximum peak amplitude Δp_{MAX} of the inclined "flat top" had only a small effect. The results for equal loudness were well fitted by an empirical law $\Delta p_{SH} + 0.11\ \Delta p_{MAX} = \Delta p_N$. This shows that for equal loudness the peak amplitude Δp_{MAX} of the flat top signature is substantially higher than that of the N-wave; thus for equal amplitude the flat top signature is the quieter.

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CONTENTS

Continued.....

1. INTRODUCTION

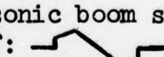
It is increasingly evident that persistent or intense noise has a complex and often harmful effect on human beings. This is especially true in the case of sharp transient or impulsive sounds. Examples are the noise of jack-hammers, industrial percussive processes, quarry blast waves, gunshots, and sonic booms. Relevant research on such noises has been both ad hoc and scientific; some major objectives have been (i) to find valid methods for predicting the loudness of a given transient, given its waveform or spectrum; (ii) to establish damage-risk criteria; (iii) where practicable, to reduce the amplitude at the source, or to modify its waveform for reduced loudness (e.g., as in recent sonic boom research).

There do exist several proposed methods for predicting the subjective loudness of sonic boom signatures (e.g., [1], [2]). Being based on specific spectra, these have the potential for application to arbitrary impulsive wave forms. The aim of the present investigation has been to test, and revise if necessary, these predictive methods in application to a variety of impulsive waveforms.

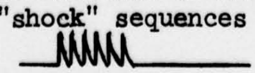
A loudspeaker-driven simulation booth ([3]-[5]) with extended rise-time capability (down to 0.22 ms) has been used for subjective loudness judgements for comparison with the theory. A new computer-aided scheme was developed to predistort the electrical input signal to counteract the loud-speaker-booth distortion. The essence of the idea is to alter the electrical input spectrum by the inverse of the loudspeaker-booth transfer function (complex frequency response), thus effecting cancellation.

Four series of tests were made, each of interest in its own right, aside from providing a predictive test. Three of these had to do with sonic boom waveforms - standard and nonstandard - and a fourth with simulated quarry blast waves.

The first series deals with the standard sonic boom, with pressure-time signature like a letter "N". The steepness of the front and rear of the "N-wave" (which are shocks) are described by their 'rise-time', and the interval between - length of the N - by the 'duration'. The tests attempt to check earlier empirical dependence [6] of the subjective loudness on rise time, duration, and N-wave amplitude.

The second series deals with certain 'flat-top' sonic boom signatures: they look like a letter "N" with the top and bottom cut off:  According to current theory ([7]-[10]) such 'flat top' signatures would be generated by a special family of very long ($\geq$ 300 ft) SST aircraft designed for minimized sonic boom.

The third series explores the contribution of the 'infrasonic' low frequency content of sonic boom N-waves to their subjective loudness, or, alternatively, annoyance. For the test N-waves the frequencies below 25 Hz and below 50 Hz, respectively, are cut off by digital filters simulating simple RC circuits. Comparison is made with a standard unfiltered N-wave.

The fourth series deals with the loudness of simulated quarry blast waves. These are generated as multiple "shock" sequences (simulating echoes), with exponential decay between shocks:  The length (number of shock-decay repetitions) of the sequences is varied over a wide range: from short impulsive bursts to almost continuous sound.

For each of these test series, using human subjects for loudness judgements, the results were compared with loudness predictions. For the first series both the Zepler-Harel [1] and Johnson-Robinson [2] predictive methods were used. The latter appeared to have somewhat superior accuracy over the full range of rise times, as well as greater simplicity. Thus only the Johnson-Robinson method was applied in all the later work. For the fourth series - the simulated blast waves - an ad hoc modification was made to the method to bridge the transition from impulsive to continuous sound.

2. REVIEW AND BACKGROUND

2.1 Introduction - Effects of Noise on People

A commonly used definition states that noise is unwanted or unpleasant sound. Thus noise is another pollutant in the human environment. It can cause auditory problems by masking speech; further, it can produce "extra-auditory" effects by disturbing physiologic functions. It can also change task performance, disrupt rest, relaxation, and sleep. People exposed for long periods of time are complaining more often that noise-induced stress has a degrading effect on their health. The reported problems (e.g., Welch and Welch [12]) include neurologic, digestive, and metabolic disorders, cardiovascular problems, and even mental problems, hypertension, nervousness, etc.

It is well known that high intensity sound may produce temporary or even permanent losses of hearing. This is documented by a large number of cited examples and published investigations. The main task of those investigations was the production of a damage risk criterion (e.g., Ward [13], Walker [14], Rice and Martin [15], etc). The necessity of better understanding of these complex problems helped motivate the present study of the loudness of impulsive sounds.

2.2 Mechanism of Hearing

The perception of sound by the human ear is a complicated process that is understood in the main, but has aspects that remain controversial.

The ear consists of three main parts: the outer ear which "matches" the impedance of the ear-drums to the air, the middle ear which transmits the mechanical vibrations of the ear-drum to the third part called the inner ear. Inside the inner ear along the basilar membrane of the cochlea the sounds are analysed and transferred to the nervous system.

The human hearing mechanism acts together as a microphone, a highly selective frequency analyzer, a sound localizer, and, via the neurological system and brain, an interpreter of the loudness, the pitch, and the timbre of sound. The acoustic pressures over which the human ear can operate cover the range from 1000 microbars down to 0.0001 microbar, which is extremely wide. The frequency range covers about ten octaves, or approximately from 20 to 20,000 Hz for youthful, healthy ears.

The oldest and perhaps most comprehensive theory of auditory analysis was developed by Helmholtz in 1863 [16]. Since then, however, the Helmholtz assumptions were proved doubtful and many newer theories have been developed. Helmholtz assumed that the ear separates a complex sound into sinusoidal components corresponding to those in a Fourier analysis and that every discriminable pitch corresponds to one particular group of nerves. These groups were supposed to be connected to the specific cochlear segments resonating to the specific tones. One of the problems not satisfactorily solved by this theory refers to the response to several simultaneous tones, as in a chord: this is heard as a single complex tone, and is not decomposed into the individual pure tones. It is also hard to explain the behaviour of the cochlear "resonators" in Helmholtz' model.

In this century, von Bekesy (e.g., [17], [18], [19]) stands as a giant in auditory research. Using surgical and physical methods on test animals, he advanced very far in clarifying the role of the tapered helical organ called the cochlea. By means of a mechanical model he gave credibility to the notion of the spatial mapping of frequency response along the cochlea ("place theory"): the lower the frequency, the closer the point of peak vibratory response to the apex (small end) of the cochlea.

In general, two "frequency theories" explaining pitch discrimination were developed. The first one assumes that in the low and middle ranges of the tonal spectrum, pitch is determined by the frequency of the neural impulses; the second one assumes existing preliminary mechanical analysis at the basilar membrane. Both of the classical theories (i.e., frequency and place theory) presuppose that the frequency analysis is completed before any action in brain can take place, and both have many theoretical problems to account for.

Some researchers, like Weddell [20] and Pfaffman [21], in connection with cutaneous and gustatory senses, produced evidence that the discrimination process of sensation quality depends on activity in a group of nerves rather than in one nerve fiber. These findings support modern views on auditory theory suggesting that the sensory patterns are interpreted by the brain and, therefore, postulate some central analysis (Nordmark [22]).

The general conclusion, based on today's knowledge, is that there is no simple and unique relationship between physical parameters of sound and human perception of sound. From the subjective point of view the physical characteristics of sound, such as intensity, frequency, etc., may be described in terms of three basic subjective characteristics, i.e., loudness, pitch and timbre. None of them is dependent strictly upon a single physical quantity, though it is possible to connect each of them primarily to some physical characteristic of sound.

The sensation of pitch is determined mainly by the frequency of the sound, but it is also a function of the intensity and wave-form. Surprisingly, doubling the frequency does not double the pitch.

Timbre is a subjective characteristic, which makes it possible to distinguish between two tones having different wave-forms, even when the intensity level and fundamental frequency are the same; but it is also a function of intensity and frequency.

Loudness is determined by the intensity of the sound and its spectral distribution.

2.3 Loudness of Sound

2.3.1 Definition and Measurement of Loudness

The first significant attempts to define and measure the subjective quantity of sound, associated with magnitude, were made by Fletcher and Steinberg [23] in 1924 and later in 1925 by Steinberg [24]. In 1957 Fletcher and Munson [25] defined loudness as the "magnitude" of sound and established a 1000 Hz pure tone as a standard tone against which the loudness of other sounds can be judged.

Stevens in 1936 [26] introduced the widely used loudness scale with a unit called sone, which is defined as the loudness of a 1000 Hz pure tone at a sound pressure level of 40 dB (re 0.0002 microbar). The doubling of the sound loudness gives a double value on the sone scale. It was mentioned before that loudness as a subjective characteristic of sound depends upon the intensity of the sound and also is a function of its frequency spectrum. Various investigators determined equal loudness contours for pure tones and for band noise. Fletcher and Munson [25] determined the sound pressure levels of pure tones required to be judged equally as loud as a 1000 Hz reference tone over most of the auditory frequency range. This and other studies are not entirely in agreement. A consensus was arrived at by the International Organization for Standardization in the form of standard ("normal") equal-loudness contours for pure tones [27]. The ISO recommendations are commonly used as the statistical result for the total population with the restrictions for the position of a source, free field conditions and plane wave sound, binaural listening, etc.

Recently there have been a few attempts to extend the existing equal-loudness contours toward the lower so-called "infrasonic" frequencies (e.g., Whittle et al [28]).

One of the most important problems in loudness research has been the development of a method for theoretical prediction of its level for the complex sounds. First attempts at a loudness calculation based on the frequency spectral measurements produced very complicated and therefore not very useful procedures (Steinberg [24], Fletcher and Munson [29]). Two of the newer ones, capable of dealing simply with broad band continuous spectra, were adopted by the International Standard Organization and are widely used [30].

From his original procedure [31], [32] Stevens developed a new method called the Mark VI [33] for predicting loudness levels of noise measured in octave, one-half and one-third octave bands. This procedure adds the fraction of so-called "loudness indexes" in the bands to the "loudness index" of the loudest band:

$$\text{Loudness} = S_m + f(\Sigma S - S_m)$$

where S = sum of loudness indices of all bands,

S_m = greatest of loudness indices in any one band,

f = fractional portion dependent on bandwidth.

Values are found from the graph, which is slightly different from the equal-loudness (sone) contours and was found by successive approximation (Stevens [33]). The result of the calculations is obtained in sones; however the loudness is very often expressed in logarithmic terms, called "phons". The phon is equivalent to the decibel, but instead of the usual definition, similar to the decibel definition (20 $\log_{10}$ of the ratio of two loudness levels) it is derived from psychological units. The loudness of any tone on the same equal-loudness contour has a value in phons which is equal to the sound pressure level in dB of the 1000 Hz tone on the same equal-loudness contour. An increase in loudness level of 10 phons is approximately equivalent to doubling the subjective loudness (in sones) in the range of mid and high levels.

In 1971 Stevens published a report with a new modified procedure called Mark VII [34], which gives the perceived level of loudness or noisiness in "PLdB" units. The new different standard reference sound is used, and the procedure allows the possibility of calculating the perceived levels of loudness for transient sounds (sonic-boom), by incorporating the Johnson-Robinson procedure [2].

The ISO also endorsed a procedure developed by Zwicker [35], [36]. Together with Flottorp and Stevens [37], Zwicker demonstrated the existence of 24 so-called "critical bands" in the audible frequency range. Beyond each critical bandwidth any further increase in a band of noise was found to have no influence on the amount of masking produced by that noise on a pure tone at the centre of a band. Zwicker assumed the existence of a functional correspondence between masking and loudness, and on this basis developed a method for calculating the loudness of complex sounds. He prepared several graphs on which the vertical divisions are marked in sones and the horizontal divisions divide the scale according to the equal critical bands in Hz (approximated by one-third octave bands). Plotting the sound spectrum on such a prepared graph and drawing in the lines for spread of masking encloses an area which is proportional to the total loudness of the sound. Zwicker defines as 1 sone the area encompassed on the graph by a one-third octave band centred at 1 kHz at a sound pressure level of 40 dB (including the correction for masking). The Zwicker and Stevens methods give estimates of loudness levels for the same sounds, differing by approximately 3 to 5 phons.

It is worth mentioning, at this point, the existence of another response concept called perceived noisiness. The perceived noisiness is defined as a subjective impression of the annoyance or unacceptability of a sound. The units of the noisiness are called "noy" and "PNdB" in analogy to sone and phon.

2.3.2 Loudness of Transient Sounds

In a previous paragraph we discussed the general concept of loudness in the context of steady-state sounds. In the case of transient sounds the situation becomes much more complicated.

Numerous experiments have established that the ear responds to sound energy as integrated over a certain time. The loudness of the short pulses grows with their duration, and some evidence shows that the loudness level (which is logarithmic) is nearly directly proportional to the logarithm of the duration, for sounds with narrow-band spectra. In general we can regard the whole auditory system as an almost linear energy integrator for transient sounds; however some experiments have shown different behaviour (Miller [38], von Port [39]) at levels near the threshold of hearing.

An early assumption concerning temporal integration was that the ear might perform as a perfect integrator and so would show 3 dB reduction of the auditory threshold for each doubling of a signal duration (Garner and Miller [40]). The very short signals as well as very long ones were found to deviate from this simple rule; therefore there were attempts to describe the relationship with a three line graph (Green et al [41]). Harris et al [42] in 1958 introduced "critical duration" defined as the intersection of the most linear portion of the integration function with the abscissa $10 \log(I_t/I_\infty) = 0$ where I_∞ = threshold intensity at $t = \infty$.

In 1959 Plomp and Bouman [43] derived two hypothetical response functions based on a simple electrical model; they found that the following equation provides a best description of their experimental results obtained near the threshold of hearing:

$$10 \log_{10}(I/I_{\infty}) = -10 \log_{10}\left(1 - e^{-t/\tau_c}\right) \qquad (2.1)$$

where τ_c = time constant,

t = duration of tone pulse,

I = threshold intensity of tone pulse,

I_{∞} = threshold intensity at $t = \infty$ (sustained tone).

Zwislocki [44] developed a quantitative psychophysiological theory which gives an explanation of the phenomenon of the loudness level's dependence upon the stimulus duration. His theory is based on the evidence that the apparent temporal summation of acoustic energy is a result of a neural summation at a high level of the auditory system.

Due to the large differences in empirical measurements caused by the lack of accurate experimental conditions, different types of stimuli used in experiments and large individual differences between subjects, there is considerable disagreement about the value of the time-constant of the ear. The reported results range from 20 to 200 ms, which can produce up to 10 dB difference in loudness. The first investigations of this problem were done by Bekesy [17] in 1929, who obtained the value of the constant equal to 180 ms, and later by Munson [45] with the result of 250 ms.

Von Port [39], using gated noises of various bandwidths, centre frequencies and levels averaged the results of the measurements to 70 ms.

Niese [46], summarizing a number of studies, obtained the average value of 25 ms. Other experiments gave different results: Stevens and Hall [47] about 150 ms, Zwicker [48], [49] 100 ms, Small [50] from 10 to 20 ms, etc.

In all cases the ranges of judgements between individual subjects were reported to be very large, which probably reflects the difficulties of the judgements themselves or substantial individual variance.

Some of the investigators have found the value of the time-constant to be dependent on a sensation level (Small et al [50]), while others did not obtain any indication of such dependence (e.g., von Port [39]). The controversy whether the time-constant of the human auditory system depends upon the frequency of a stimulus is also not resolved. The loudness of transient stimuli compared to continuous sounds depends also upon other characteristic parameters describing the waveform and its duration. The rise time and repetition rate are strongly correlated with the loudness level of the impulsive sounds.

The influence of rise time on the loudness has been studied by Carter [51] who used triangular pulses with very short rise times (0.05 - 0.5 ms). Gjaevenes [52] used tone bursts with rectangular envelope and rise times between 30 and 950 ms. From both experiments authors concluded an increase in loudness with a decrease in rise time. Carter suggested that the relationship between the loudness level and the rise time is of a logarithmic type. Gustafsson [53] used in his experiments pulses from a transient sound generator having rise times from 0.3 to 10 ms. Results showed the connection between loudness and rise time in a form of a power function, which tends to confirm Carter's results.

The effect of repetition rate on the loudness of short pulses was also studied by Carter [51]. He presented data showing that for the triangular transients with 0.5 ms rise time and 1 ms duration, having the repetition rate from 1 pps (pulse per second) to 128 pps, the doubling of the repetition rate gives 3 dB rise in loudness of the signatures. A repetitive train of noise bursts (say 10 ms on and 10 ms off) is judged to be about 3-5 dB louder than a spectrally similar continuous noise of equal peak level (Pollack [54]); however, the continuous sound contains about 8 dB more energy. Zwicker [48] showed that this effect can be replicated when subjects are instructed to judge annoyance but he found no such effect when they were instructed to judge loudness.

2.4 Human Response to Sonic Boom: Loudness

2.4.1 Introduction

Sonic-boom research involves a whole spectrum of different fields and problems. The research in physiology covers sleep losses, thresholds of waking and stress-induced responses of the human beings and animals. From a psychological point of view startle measurements and task interference are the main subject of the investigations. Group actions, complaints and community protests are part of the sociological programs.

The chosen environment and investigative method in sonic-boom research, as well as in research involving other types of noise, often depends upon the purpose of an experiment and problems connected with it. In laboratory studies with simulated or reproduced signals, stimulus parameters and the investigative environment situation can be easily controlled. The most publicized sonic-boom experiments, however, have been carried out under field conditions, using actual aircraft overflights. Such relatively uncontrolled field studies are the only way to do community research.

2.4.2 Simulation Techniques

The laboratory research on sonic boom usually requires long and very extensive programs with a possibility of controlling and easily varying the signature parameters. Therefore, it is necessary to find other ways of producing test signals than the very expensive and inconvenient supersonic overflights. With this motivation various types of simulators have been developed in a number of laboratories. They are usually highly specialized and based on different principles of operation, due to specific research requirements.

One scheme of sonic boom simulation utilizes a tiny precision model of an SST placed in a wind tunnel. Projectiles fired in a range are also used. Propagation experiments may be conducted with these simulators.

Shock-tube or compressed air driven simulators are used for investigating propagation phenomena as well as sonic boom response of humans, animals and structures. The driver is coupled to a large pyramidal horn down which the travelling N-wave propagates. The driver device may be a shock tube with diaphragm (Warren [55], Slutsky and Arnold [56]) or a specially designed compressed air value (Tamboulian and Peschke [57], Glass et al [4]). A shock tube with a very short horn formed the basis of a portable sonic-boom simulator for field studies of wildlife (Gottlieb [58]). For large-scale outdoor environmental experiments, sonic-boom waveforms may also be simulated by multiple or specially shaped (for better control) explosive charges.

Another type of simulation facility employs a piston as driver. In the NASA Langley Research Center the low-frequency piston facility, with electrohydraulic drive and cylindrical test chamber were employed for structural and environmental testings by Edge and Mayes [59]. Similar facilities at the Wright-Patterson Air Force Base and at the Stanford Research Institute, have been used specifically for testing single human subjects (Lukas and Kryter [60]).

However, the most convenient and flexible technique for the experimental testing of the human response to the simulated sounds (e.g., sonic-boom) employs earphones or loudspeakers for the simulation. In order to obtain the required low frequency response (down to DC in case of the sonic-boom spectrum), it is necessary to seal either the earphone or the loudspeaker-driven chamber to be virtually airtight. Additionally a specially equalized electronic amplifier must be provided. Sonic boom subjective experiments with the application of earphones were conducted by Thurner [61], Ellis [62], Zepler and Harel [1] and later by Rood [63] at the University of Southampton. This type of facility requires the development of a device which will secure the airtight fit of earphones to the head of a subject. However, the main disadvantage of the earphone system is the lack of the whole body exposure to the sonic boom pressure field, an exposure which might be important for some subjective studies.

The airtight loudspeaker-driven chamber overcomes this deficiency. In 1965 Pearson and Kryter [64] used a 100 ft^3 chamber with 18 inch loudspeakers mounted in a wall and driven by an amplifier. The signals were generated by means of the device giving the desired electrical waveforms by following a silhouette placed in a "photoformer". Also in 1965, a similar chamber facility was developed and built in the Bioacoustic Laboratory at Lockheed for research by Shepherd and Sutherland [6]. The loudspeakers in this facility were mounted on the door and were driven by high power DC amplifiers employing special equalizing circuitry. The input test signals were recorded on a magnetic tape. An electronic noise squelch circuit was employed for reducing the background noise. The 70 cubic foot chamber allowed overpressures of 4.5 psf for long rise times (10 ms), and 2-3 psf for the shortest rise time of 1 ms.

The simulation device developed in the University of Southampton in the form used by Rood [63], consists of the 120 cubic foot chamber and an N-wave electronic generator. The additional frequency compensating network and negative feedback system allowed the attainment of 1 psf overpressure at 2 ms rise time.

The current investigation was carried out in the loudspeaker-driven simulation booth developed at the Institute for Aerospace Studies, University of Toronto, which is described in Chapter 3.1. Building from the experiences with the earlier booths, substantial advances in design have been incorporated. Rise times as low as 0.2 ms are routinely obtained with overpressures of order 1 to 2 psf, and good waveform simulation.

2.4.3 Prediction Methods for Loudness of Sonic Boom

In Chapter 2.3.1 we discussed methods for calculating the loudness of continuous sounds. Now we will concentrate on the methods specifically developed for calculating the loudness of impulsive sounds.

In 1963 von Port [39] suggested a procedure which transformed the energy of the impulsive sounds to such a form that it was possible to apply the equal-loudness contours for steady-state sounds in the prediction process. He assumed that the ear integrates the energy over a 70 ms interval. The value of the critical time was found through subjective measurements of the loudness of narrow and broad bands of white noise pulses as a function of duration. For pulses shorter than 70 ms the measured sound energy was divided by 70 ms and the loudness was calculated as for a continuous sound. For repeated impulses von Port's procedure is more complicated and involves a different decay time (350 ms).

Niese [46] in 1975, using the same basic ideas as von Port, proposed a method which differs mainly in how the steady-state measurements should be made. The method starts with the A-level weighted calculation, corrected by the size and frequency of the largest one-third octave band measurement. The integration time used by Niese is 25 ms.

Kryter [65] attempted to develop a mechanistic calculation procedure in which one calculates "perceived noise level" (PNL) over 0.5 ms intervals. For impulsive sounds the calculations were corrected for "startle".

All the procedures mentioned above were designed to calculate the loudness of the impulsive sounds in general. The two following methods were developed with the particular purpose of calculating loudness of the sonic-boom signatures.

Zepler and Harel [1] concluded from the various experimental data that the mechanism of critical bands for impulsive sounds is not clear. There are some indications that the critical bands are created by the signals themselves or that the parameters of the mechanism of frequency selectivity vary under intelligent control. Therefore, they implied that it is not possible to measure such critical bands for short pulses. In their research, Zepler and Harel have tried to find a relationship between weighted (in relation to the equal-loudness phon contours), energy of the stimulus and the subjective loudness of a sonic-boom signature. They calculated so-called "weighted energy density" $k|F(\omega)|$ of the sound. The factor "k", which converts decibels into phons, is chosen with some arbitrariness. Based on the empirical evidence, the authors assumed that the unit step function (at 127.6 dB peak pressure) at the frequency of 50 Hz affects the hearing in the same way as 50 Hz continuous tone at the level of 80 dB SPL. At 1 kHz the value of the factor "k" is unity.

The general relationship is as follows:

$$k = 10^{n/10}$$

where "n" is the difference between phons and decibels at the appropriate level, for a particular frequency. In their report, they suggested that the reference level chosen for the calculations [1] may not be best fitted and Rood [63] has compared his experimental results with theoretical calculations based on this method, for a few reference levels, and found that the best results were obtained for a 100 phon weighting network. The chosen level determines the weighting factor "k" The subjective loudness is proportional to the area under the weighted energy density curve.

The Zepler and Harel method for predicting loudness was also supported by experimental evidence by Rice and Zepler [66], for short duration waveforms caused by firing a pistol shot.

The second sonic-boom loudness prediction theory, by Johnson and Robinson, was developed in two papers, published in 1967 [67] and in 1969 [2]. It follows von Port's proposal in many respects. The procedure is based on the assumption that the ear integrates the energy of a sound arising in a given frequency band completely if the whole of the energy is received within less time than the auditory time constant. Thus, the loudness of the short transients is determined by the sound energy, not by the sound pressure level, in the spectrum bands. The value of 70 ms, as a compromise between the growth and decay time constants of the integration process, was chosen.

The energy spectral density $|F(\omega)|^2$ of the transient stimuli may be obtained through digital Fast Fourier Transform procedure or any other method suitable for the short sound pulses. The Fourier transform of the "ideal" N-wave signature determined analytically is given by the equation (Ref. [2]):

$$F(\omega) = \int_0^{T+\tau} p(t)e^{i\omega t}dt = i4\Delta pT \left(\frac{\sin\frac{\omega T}{2}\cos\frac{\omega\tau}{2}}{\omega^2(T+\tau)T} - \frac{\sin\frac{\omega\tau}{2}\cos\frac{\omega T}{2}}{\omega^2(T+\tau)\tau} \right) \quad (2.2)$$

where T = duration of the signature,

τ = rise time of the signature.

In order to simplify the computations, Johnson and Robinson proposed that the energy in the one-third octave bands may be calculated from the envelope of the spectrum, since for the frequency range of interest the actual spectrum oscillates rapidly (rate 1/duration) within the one-third octave bands (see Fig. 1). The envelope of the spectrum function is defined by:

$$\frac{T^2}{\omega^4}\sin^2\frac{\omega\tau}{2} \quad (2.3)$$

because the maxima of the function from the equation (2.2) occur at the frequencies $\omega = 2\pi n/T$. The area under the envelope represents approximately twice the real band energy, therefore the energy in each one-third octave band is proportional to:

$$\frac{16(\Delta p)^2 T^2}{\pi(T - \tau)\tau^2} \int_{\omega_1}^{\omega_2} \frac{\sin^2 \frac{\omega\tau}{2}}{\omega^4} d\omega \qquad (2.4)$$

for an N-wave.

In the attempt to relate the impulse energy spectral density to the steady-state sound loudness calculation, Johnson and Robinson defined the "effective band pressure levels" of sonic boom, as to be equal to the levels of a continuous steady noise which over a period equal to the critical time contain the same energy. This is obtained by dividing the energy spectrum density levels (calculated directly from the waveform by the Fourier transform, or produced in analog form by the spectrum analyser) of the transient sound by the assumed critical time. The method is valid only for pulses shorter than the critical time. The resulting quantity has the dimension of the sound power and can be expressed as a sound pressure level in decibels, suitable for loudness calculation by any of the existing procedures for continuous sounds (see Chapter 2.3).

The condition for the pulse duration refers to the effective duration of the response to the stimulus at the level of the cochlear frequency-selective mechanism and not to the physical duration of an impulse. Johnson and Robinson have estimated [67] that, for typical sonic bangs with duration from 100 to 250 ms, even if the response for the lowest frequency bands is longer than 70 ms, the error will be less than 1 dB. This is based on the assumption that the human frequency-analytic mechanism has the properties approximated by a set of the one-third octave filters at the higher frequencies and similar set of filters with wider bandwidth than one-third octave at the lower end of the audible spectrum. It can be assumed that for these filters the significant part of the response for the brief pulse is of the order $1/\Delta f$ (Δf-bandwidth), which means that the sonic-boom N-wave signatures are perceived by the auditory system as two separate, individual pulses, roughly less than 100 ms long for all bands below the 50 Hz band, approximately.

Having the one-third octave band pressure levels in decibels (re 2×10^{-5} N/m^2), a few corrections are introduced before applying the loudness calculation procedure. First, because the duration of the typical sonic-boom signature is long enough so that the two bangs (front and rear) are heard separately, the calculated band pressure levels are reduced by 3 dB. It is not clear, however, what the limitations for this correction are in the sense of the range of durations for which it should be applied. Secondly, in order to take into account the high energy levels at very low frequency (e.g., 3-4 Hz), each band pressure level below the 50 Hz one-third octave band is reduced by means of the equal-loudness contours to that level at 50 Hz which produces the same loudness level in phons. These weighted levels are then combined with the existing level in the 50 Hz band by simple decibel addition. The data prepared this way may be used for loudness evaluation based on any standard procedure. Johnson and Robinson have recommended the Stevens' Mark VI procedure [33], because of its great simplicity and the good results achieved by its use.

Although illustrated in the foregoing for sonic-boom N-waves, the Johnson-Robinson predictive method has a much broader affiliation: it may be used to calculate the loudness of arbitrary impulsive waveforms. The step-by-step handbook procedure for our computer implementation is detailed in Appendix 8.

2.4.4 Investigations of Sonic Boom Loudness: Review

A few years ago research into the acceptability of the sonic-boom by human beings was mainly concerned with possible temporary threshold shift (TTS) or permanent threshold shift (PTS) in hearing response. It was concluded, however, that these were negligible, so that the sonic-boom can be disregarded as a threat to the auditory system (e.g., von Gierke and Nixon [68]). Today the primary direction of the research is concentrated on the impressions of loudness and annoyance.

Field studies of sonic-boom have been conducted extensively in the USA, England, France, and Sweden. Comparative investigations of loudness and annoyance for sonic-boom versus jet aircraft flyover noise were carried out in Great Britain in Project Westminster [69] and in the USA in the Edwards AFB Program [70]. In both cases the observations were made indoors and outdoors. A representative result may be quoted from the flight tests at Edwards AFB, ". . . Forty percent of those polled rated booms of about the level projected for current supersonic transports in cruise (20 psf) as unacceptable for outdoor listening; for indoor listening the corresponding percentage was 30%." [71].

Johnson and Robinson [67] in 1967 conducted a series of field experiments in which 61 subjects made judgements of the relative annoyance of sonic-boom, explosions and jet aircraft noise, using the method of direct magnitude estimation. The results of the subjective experiments correlated closely with calculated loudness levels, suggesting the possibility of application of the proposed theory for different types of spectra. The correlation with perceived noise and peak overpressure was poor. In their later paper [2] the prediction theory for sonic-boom was used in theoretical calculations of loudness level as a function of rise time, duration, and delay time (between the incident and the ground-reflected waves). The authors assumed, taking into account previous work, that the difference between the spectrum of the real recorded sonic-boom and an idealized approximation consisting of up to seven straight line segments are of little consequence in determining loudness. Therefore, the calculations were made for the analytically derived Fourier transforms of the idealized N-wave signatures, using the proposed prediction procedure. The results show no variation in the loudness within 0.1 phon in the duration range from 100 to 500 ms for an N-wave having 0.1 ms rise time and zero delay time. The change in rise time at ground level (i.e., delay time equal to zero) from 16 to 0 ms gives the increase in loudness of 25 phons. Within the range from 16 down to 4 ms loudness rises at a rate of about 3/4 phon per millisecond increase of rise time, and below 4 ms the increase is even sharper.

The influence of delay time (between the direct and ground-reflected waves) depends upon the rise time value, but in general the loudness declines with increase of delay time.

Turning now from field studies to laboratory studies, at the University of Southampton research into the subjective effects of the sonic-boom has been carried out by Ellis [62], Zepler and Harel [1] and Little [72] using earphones, and by Rood [63] using a loudspeaker-driven booth.

Zepler and Harel in their subjective tests used specially designed earphones developed previously by Thurner [61]. The frequency response of earphones was contrived to be practically flat between zero and 1.5 kHz. This flat frequency response allowed reproduction of N-wave signatures with relatively good quality, but the performance depended strongly upon air leakage which was hard to avoid. The maximum pressure obtainable was approximately 2 lb/ft^2.

The comparison technique was used, with a continuous tone of 400 Hz as a reference signal. While this reference stimulus has the advantage of giving results directly related to the definition of loudness units, it was reported to be very difficult for the test observers to compare loudness of such extremely different signals. Altogether ninety subjects took part in the experiment and the standard deviation for the last series of the tests for 20 subjects was reported by Zepler and Harel as being about 4.5 dB. Despite the large discrepancy between subjects, results for individual subjects were usually consistent within a few decibels.

They found a decrease of the loudness level, at the same overpressure, equal to 16 dB within the rise time range from 0.3 to 10 ms. This result was confirmed by theoretical calculations based on the new proposed method. Both the experimental and theoretical estimations have shown that, for rise times larger than 1 ms, variations in rise time at constant overpressure had more effect than the corresponding variation in the overpressure at constant rise time.

The authors performed an additional test with high pass filtered N-wave signatures showing no loudness effect for cut-off frequencies lower than 40 Hz, for N-waves having rise times of 1 ms and 3 ms. They also reported that when the duration of the N-wave signature is shortened to 50 ms or less, the <u>two</u> bangs heard for longer signatures separately merge into one one with an <u>increase</u> of about 3 dB in loudness.

In 1973 Rood [63] published a Ph.D. thesis in which he extended Zepler and Harel's research. He measured the absolute subjective loudness of half N-wave signals, using a loudspeaker-driven booth as a simulation facility. The pressure booth enabled him to produce signals with rise times from 16 down to 2 ms and overpressures from 0.5 to 3.0 psf (the latter for the longer rise times). The experimental technique involved the comparison of N-waves with 200 Hz and 400 Hz tone bursts, 20 ms long. In his paper Rood reported some difficulties in finding the effective pitch and subjective duration of the N-wave signature to be matched by reference signals. A comparison between tests with earphones and the pressure booth was also conducted, but no significant difference in respect to subjective loudness determination was found.

Rood's experiments were carried out for three different overpressure levels: 0.5 psf, 1 psf and 2 psf. The loudness comparison test results were converted into the absolute loudness level scale through the experimentally established relationship between the reference tone bursts and 500 ms long continuous sound and these values were compared then with the theoretical predictions. The standard deviation varied from 2.44 to 7.16 dB. The best fit to these results was obtained by the Zepler prediction procedure, for 100 phon weighting curve. In general, the change from 2 ms to 16 ms in rise time caused a decrease of subjective loudness level of 17 to 19 phon, depending upon the peak pressure. It was found that while the doubling of the overpressure results in an increase of loudness by 6 phons for longer rise times, it is marginally less for shorter rise times.

On comparing his results with earlier works by Zepler and Harel [1] and Ellis [62], Rood concluded that the previous results underestimated the true loudness levels by 12 phons because of the test method, which involved comparisons of the impulsive sounds with the continuous sound reference.

The most complete and significant experimental work was done in 1968 by Shepherd and Sutherland [6] at the Stanford Research Institute. They employed the Lockheed loudspeaker-driven sonic-boom simulation booth (see Chapter 2.4.2). The total subjective test series was divided into three parts; in each the paired comparison technique was used. One boom in each comparison pair was designated the standard, the other being the boom under test; the relative amplitudes were adjusted for equal subjective loudness. The relationship between relative amplitude versus rise time and duration was defined at three different amplitude levels. Values of the parameters of sonic-boom signatures used in the experiments ranged from 1 to 10 ms for rise time, from 100 to 500 ms for duration and from 0.8 to 2.4 psf for overpressure. Similar sets of comparison tests were repeated for annoyance as a judgement criterion.

In the second part of the experiment the subjects evaluated the loudness of half N-wave signatures with additional sawtooth serration; in the last part there was an additional sawtooth having different interpeak spacing. Shepherd and Sutherland's basic results indicate no significant change in loudness (inferred from amplitude ratio in dB) of the half N-wave with respect to duration while the increase of the rise time from 1 ms to 10 ms was followed by a decrease in loudness of 13 dB.

Increasing the amplitudes of the two signals of a pair in the same proportions had no effect on the relative loudness. Also the difference between loudness and annoyance judgements were insignificant, except for those involving signatures with 10 ms rise time.

The addition of a spike to the half N-wave to bring the amplitude to 3.3 psf was tried; it provided around 8 dB difference in loudness, when compared with the half N-wave signature without spike, of overpressure 1.6 psf.

There was no significant difference in loudness between signatures with added sawtooth, so long as this additional fine structure did not change the general shape of the waveform and modify its rise time.

3. EQUIPMENT

3.1 UTIAS Sonic Boom Simulator - General

In order to obtain more significant and precise data on human response to sonic boom and other types of noise, the development of a loudspeaker-driven booth was initiated in 1970 at UTIAS. A series of modifications and improvements led to a capability of faithful simulation of the sonic boom N-wave and other transient waveforms. A particular feature was the attainment of rise times as short as 0.2 ms at an overpressure of 1 psf (longer rise times at higher overpressures up to 4 psf). This rise time is five- to ten-fold shorter than that of the earlier cited facilities (Shepherd and Sutherland [6], Rood [63]).

The UTIAS Sonic Boom Simulation Booth (Fig. 2) consists of an airtight 2.1 m^3 volume chamber driven by 12 loudspeakers in two sizes mounted in the aperture of the wall faced by subjects. The booth features a double-wall plywood construction with inside wall surfaces heavily lined with sound absorbing semirigid fiberglass material to minimize high frequency reflections and consequent resonances; the free-air volume is thus reduced to about 1.3 m^3. The absorbing material was chosen by taking into account easiness of attachment, porosity, absorber thickness and absorption coefficient.

An airtight window was installed on the door to allow visual observation of the subject seated inside. In the original form the ventilation system was built in, which allowed a very slow flow of fresh air into the booth during operation. It was found impractical, however, and was later removed to minimize air leakage. The booth is also equipped with an internal door release mechanism and locking system, intercom and a support system for the microphone.

3.2 UTIAS Sonic Boom Simulator Electronic System

The UTIAS Sonic Boom Simulator employs six 15 inch low-frequency loudspeakers (Altec Lansing woofers, model 515B) and six 8 inch medium-frequency loudspeakers (Radio Shack 30 watts, type 40-1286 with the frequency range up to 20 kHz) to cover the total desired frequency range from about 0.1 Hz to 5 kHz. The two types are displaced in their mounting in such a fashion as to equalize the effective travel time to the subject's head. Thus the signals arrive in step, so that the wave front of the composite is virtually as sharp as that from a single medium-frequency loudspeaker.

The electronic system consists of a crossover circuit, an equalizing network and four 100 W power amplifiers. The block diagram of the total system is shown in Fig 3. The low and high frequency part of the input signal are separated by the crossover circuit. The adjustable equalizing network compensates for the major part of the speaker and booth coloration of the frequency response and hence eliminates much waveform distortion. (The block diagram of the equalizing circuit is shown in Fig. 4.) The main part of the equalizing network is a two channel Altec Lansing "Acousta-Voicette" equalizer, Model 729A, containing twenty-four one-third octave filters centred at frequencies from 12,500 Hz to 63 Hz. Each filter is adjustable over a range ± 12 dB. To carry on below 63 Hz it was necessary to include a series of low pass filters in parallel in the low frequency channel (the Altec unit cuts off below 8 Hz). The total signal in this channel is summed in a special circuitry.

Careful adjustment of these filters in both high and low frequency channels made it possible to obtain a relatively smooth frequency response of the total system. This is supplemented by a special noise-squelch circuit consisting of an offset eliminator and a noise eliminator system to decrease the background noise. The system is triggered, permitting the electronic network to be switched off during the silent intervals between the test pairs.

The output signals from the equalizing network are amplified by four 100 W power amplifiers with nearly flat frequency response down to DC and less than 3 dB drop at 100 kHz.

3.3 Generation of the Input Signals

As the original input source for the UTIAS Sonic Boom Simulator, an elaborate electronic function generator was developed. This analog facility had the capability of fitting, within limits, an arbitrary voltage-time signature by means of 100 straight-line segments with slopes adjustable by individual knobs. The adjustment turned out to be very laborious; but worse than that the signatures were not stable in time because of a problem with voltage drift and residual noise (Gottlieb [58]). Thus the function generator was abandoned in favour of a computer-based approach.

The test signals used in the current project were generated from mathematical formulas by the HP 2100A digital computer with 24k memory, magnetic tape unit and conventional D/A and A/D equipment. The basic block diagram of the generating sequence is shown in Fig. 5.

A simple FORTRAN program generated the series of test pairs each having a different, randomly presented, overpressure ratio - according to the supplied data. The whole series was recorded on magnetic tape in digital form, and then after D/A conversion recorded in analog form (Bruel and Kjaer two channel FM tape recorder type 7001). The program working with the D/A converter also generated the triggering signal recorded on the second channel of the tape recorder.

As mentioned before, the equalizing network can, within its limitations, reshape the frequency response of the simulation system. However, the adjustment is not sufficient to completely eliminate distortions introduced by the system and achieve a flat frequency response. In a case of more complex waveforms (e.g., minimized sonic-boom, "flat-top" signatures), the distortions due to the non-ideal frequency characteristic of the system become more critical, significantly changing the general shape of a waveform. Therefore, a novel computer-aided method for countering the distortion was developed. Using the measured transfer function of the whole simulation system (i.e., electronic circuitry, loudspeakers, booth) the input signatures are "predistorted" in frequency domain to counter the simulator distortion. The theory behind this counter-distortion approach is given below.

Assuming that our simulation system is linear and time invariant, the output y(t) for any arbitrary input x(t) is given by the equation:

$$y(t) = \int_{-\infty}^{\infty} h(\tau)\, x(t - \tau)\, d\tau \qquad (3.1)$$

For a physically realizable system $h(\tau) = 0$ for $\tau < 0$, where $h(\tau)$ is defined as a system response to the unit impulse, applied a time τ before.

The Laplace transform of $h(\tau)$ gives a transfer function of the constant parameter linear system $H(s)$, defined as:

$$H(s) = \int_0^\infty h(\tau)\, e^{-s\tau}\, d\tau \tag{3.2}$$

where $s = \gamma + j\omega$.

The dynamic characteristics of such a physically realizable and stable system can be described by a frequency response function $H(\omega)$ which is defined as a Fourier transform of $h(\tau)$:

$$H(\omega) = \int_0^\infty h(\tau)\, e^{-j\omega\tau}\, d\tau \tag{3.3}$$

In a special case, when $\gamma = 0$ both expressions (i.e., 3.2 and 3.3) are equivalent and the frequency response function may replace the transfer function with no loss of useful information [73]. Applying the Fourier transform to equation (3.1) we get as a result:

$$Y(\omega) = H(\omega)\, X(\omega) \tag{3.4}$$

where $H(\omega) = |H(\omega)|e^{-j\phi(\omega)}$ is a transfer function of the system and $Y(\omega)$ and $X(\omega)$ are Fourier transforms of the output and input signals respectively. In the ideal case, when the system is linear and its parameters are constant in time, the transfer function $H(\omega)$ is a function of a frequency only. The usefulness of equation (3.4) lies in the fact that the transfer function concept allows one to separate the system characteristic from the input signal.

We may now represent the Sonic Boom Simulator in the form of a black box with the complex transfer function $\tilde{\Gamma}(\omega)$. If we put an ideal N-wave signature into the system, we would come up with the output signal given by the following equation (see the outline of the predistortion scheme in Fig. 6):

$$\tilde{F}_2(\omega) = \tilde{F}_1(\omega)\, \tilde{\Gamma}(\omega) \tag{3.5}$$

where $\tilde{F}_1(\omega)$ = Fourier transform of the input signal (e.g., N-wave signature),

$\tilde{F}_2(\omega)$ = Fourier transform of the resulting distorted output signal.

Knowing the Fourier transform of the input signal we can recover the transfer function of the simulating system from the spectrum of the output signal:

$$\tilde{\Gamma}(\omega) = \frac{\tilde{F}_2(\omega)}{\tilde{F}_1(\omega)} \tag{3.6}$$

The important thing in this approach is that it must be insured that an input signal, measuring the transfer function, will excite the system over the entire frequency range of interest.

Dividing the complex spectrum function of the desired waveform $\tilde{F}_3(\omega)$ by the transfer function of the simulating system, given by equation (3.6), will yield a new "predistorted" Fourier transform function:

$$\tilde{F}_4(\omega) = \frac{\tilde{F}_3(\omega)}{\tilde{\Gamma}(\omega)} = \tilde{F}_3(\omega)\,\frac{\tilde{F}_1(\omega)}{\tilde{F}_2(\omega)} \qquad (3.7)$$

After transforming back to the time domain, the new waveform $F_4(t)$ fed back into the system will counter the distortions of the non-ideal transfer function producing the desired output signal $F_3(t)$. In terms of the Fourier transform functions the total scheme can be described by the following:

$$\text{Output, } \tilde{F}_5(\omega) = \tilde{F}_4(\omega)\,\tilde{\Gamma}(\omega) = \frac{\tilde{F}_3(\omega)}{\tilde{\Gamma}(\omega)}\,\tilde{\Gamma}(\omega) = \tilde{F}_3(\omega), \text{ desired output} \qquad (3.8)$$

The above procedure has been computerized; the entire process is done by a single FORTRAN program (see Appendix 7) in the HP 2100A computer using Fast Fourier Transform hardware. The computer generated "predistorted" signatures are recorded on the Bruel and Kjaer FM tape recorder and played back into the amplifiers of the Sonic Boom Simulator in the same fashion as for simply generated signals. The simplified block diagram of the generating process is shown in Fig. 7.

From the theoretical point of view the predistortion scheme should give as a result ideally clean pressure signatures, identical to the desired waveforms, inside the booth. This is never achieved, however, in practice. Beginning with the simulator system, we made the assumption of linearity, which is only an approximation of reality. Mainly because of the speakers the simulation system exhibits some nonlinear effects; these limit the accuracy of the predistortion procedure because the effective transfer function of the system ($\tilde{\Gamma}$) is also a function of the input signal itself. The transfer function depends also upon time, which is caused by changing parameters with time (e.g., air leakage).

The discrete Fast Fourier Transform used in the predistortion procedure is in a sense inexact; it is only an approximation to the continuous Fourier transform by virtue of errors due to discrete sampling in time and truncation of the sampled signals. Each of these effects can be minimized, but the limited memory of the computer and computing time limitation forces a programmer to choose a suitable compromise (more details about FFT are given in Chapter 3.4.2).

The specific characteristics of the spectrum functions of the signals used in this investigation caused further problems for the predistortion procedure. The procedure, it will be recalled, requires the division of one spectrum by another. However, the spectra of impulsive signatures have multiple zeros (e.g., the zeros of an N-wave spectrum are spaced at intervals of 1/duration);

thus the division leads to 0/0 at these points. A number of different schemes for avoiding divisions by zero were tested; the one chosen in the final program substitutes the values for the ideal transfer function (i.e., the constant gain transfer function which does *not* distort the input signal) at points where the real transfer function cannot be defined because of the division-by-zero situation (see equation 3.6).

This arbitrary scheme for avoiding the divisions by zero may, of course, yield errors, but the problem was found to be non-critical because of the limited number of these zero points in practice (i.e., the discrete points calculated by FFT procedure may not match the frequencies of zeros). An additional means of reducing the effect of 0/0 was afforded by using the same signals for testing the transfer function as those to be predistorted, because zero points have the same position in both spectra. It also assures the proper frequency range of the testing spectrum and reduces effects of nonlinearity of the system.

All the problems discussed above introduce errors in the computed spectrum of the predistorted input signal; these are reflected in the time domain, often in the form of "ringing". In Fig. 8 there are shown some examples of the input electrical "predistorted" signals (top) and the resulting output pressure signatures recorded inside the booth (bottom). Figure 9 shows the comparison between pressure signatures recorded inside the booth, without and with predistortion. The latter figure illustrates the significant improvement in the quality of the waveforms for the predistorted signals. The remaining fine structure is due to the procedure errors.

Figures 10 and 11 show the respective energy spectrum density functions obtained through the Fast Fourier transform procedure for the ideal N-wave and for the N-wave as reproduced in the booth using predistortion. Using these functions and the Johnson-Robinson prediction procedure, the comparison between the loudness levels of both signatures was made. For the same rise time, overpressure and duration, the difference in calculated loudness was found to be about 0.4 phon. Thus the residual distortion remaining in the reproduced wave - the effect of imperfect cancellation by the predistortion process - is seen to have but a small effect on subjective loudness.

In the third part of the current project (see Chapter 5.3), high-pass <u>filtered</u> N-wave signatures were used in order to explore the contribution of the low frequencies to loudness. The signals were generated by the digital filtering of the predistorted N-wave signature in the computer. In Fig. 7 the full process of generating these impulses is shown in the form of a simplified block diagram. The N-wave signature, generated in digital form, was first transformed into the frequency domain by means of the FFT procedure and then predistorted in the frequency domain (cf above). This spectrum, after an inverse FFT into the time domain, was used as the reference signal. The same predistorted spectrum (a complex Fourier transform) was alternatively filtered by a high pass digital filter with cut-off frequency of either 25 or 50 Hz. This spectrum, after an inverse FFT into the time domain, became the test signal.

The digital filter simulated a simple, ideal RC circuit (see Fig. 12) with the following frequency response:

$$\tilde{H}(\omega) = \frac{j\omega RC}{1 + j\omega RC} \tag{3.9}$$

where RC is a time constant of the filter. The complex product of the above function and the Fourier transform of the predistorted N-wave signature $\tilde{F}_N(\omega)$ gives (after converting into time domain) a digitally calculated signature representing a signal filtered by the ideal high pass filter,

$$F_F(t) = \mathcal{F}^{-1}\{\tilde{H}(\omega)\tilde{F}_N(\omega)\} \tag{3.10}$$

The resulting pressure signatures recorded inside the booth are shown in Fig 13. They are almost identical with the theoretically calculated responses of the high pass RC filter for N-wave signature (see Crocker and Sutherland [74]). Pulses obtained by means of real analog filters are slightly different because of the non-ideal characteristic of real filters (Hilton and Newman [75]). In Fig. 14 the comparison between the calculated and analog pulses is shown.

3.4 Measurements of Transient Sounds

3.4.1 Measurements in Time Domain

Measurements of the pressure time history of the transient sounds imposes special conditions on the equipment used for this purpose. The physical parameters that describe impulsive sounds include: peak sound pressure level, rise time, rate of decay, pressure variation with time and repetition time. All of these features must be measured and recorded accurately.

Ideally a transducer and a measuring system should have a linear frequency response between DC and well beyond the highest significant frequency in the signal. However, in practice measuring equipment always has limitations in high and low bands of the frequency response, which gives as a result incorrect values of the measured parameters of the impulsive signals.

Taniquchi [76] and Hilton et al [75] have defined the requirements for the upper and lower cut-off frequency of the measuring system as the frequencies where the Fourier transform function of the shock pulse drops by 40 dB.

Crocker and Sutherland [74] defined the lower and upper limits of the equipment for measuring real sonic-booms by theoretical calculations of the filtering effects upon the ideal N-wave signature and blast waves, without taking into account the absorption of high frequencies by the atmosphere and other mechanisms which increase the rise time of the shock wave. They recommended a linear frequency response with lower and upper cut-off frequencies of 0.05 and 10,000 Hz respectively.

In the current experiment the range of the frequency spectrum for all simulated impulses used in the subjective tests extended from nearly 0.1 to 5,000 Hz; thus the requirement for equipment for measuring these impulses was an essentially flat frequency response covering that range.

The measurement system must also be stable and sensitive enough to convert the pressure variation into electrical signals which can be detected or stored on tape without degradation or contamination by noise. The transducer and electronic equipment should have: a good phase response, less than 1.5 dB ringing and should be completely damped after about 100 microsecond. The measurements should be made in a precisely specified environment in order that the experiments could be duplicated or checked by other researchers. Large nearby reflective surfaces should be avoided and the angle of incidence of the transducers should be kept constant, etc. (Ross et al [77]).

All measurements in our experiments were carried out at a fixed position of the microphone (approximately at the level of the subject's ear) in the closed, empty booth. The microphone employed was a Bruel and Kjaer 1 inch condenser microphone type 4146, with a random incidence corrector type UA 0055. This microphone is designed for sound pressure measurements down to less than 0.1 Hz (this is achieved by closing the air equalization hole of a standard condenser microphone), and the random incidence corrector makes it practically omnidirectional up to 10 kHz. Together with the microphone, the Bruel and Kjaer carrier system type 2631 was employed. The frequency range of the carrier system extends from DC to 150 kHz, and it provides an ideal first stage processing of sonic-boom and other impulses that carry a strong, very low frequency content.

The output signals from the carrier were either recorded back on the Bruel and Kjaer FM type recorder type 7003, which features the frequency range at speed 15 ips from DC to 10 KHz, or photographed by a Polaroid camera from a dual beam cathode-ray Tektronix oscilloscope with storage type 5103N. This method of measuring an impulse noise involving a microphone and oscilloscope is regarded as very accurate and is widely used in practice (Martin et al [78]).

3.4.2 Spectrum Measurements - Fast Fourier Transform

In order to calculate the loudness of transient sounds by the Johnson and Robinson procedure, it is necessary to calculate the energy spectrum of the waveforms. This was done in the computer by the Fast Fourier transform procedure, which was also used for generating test signals (filtering and predistorting procedures).

A real transient impulse may be represented as y(t) (real) in the time domain, or equally well as Y(ω) (complex) in the frequency domain. Here Y(ω) is the Fourier transform given by

$$Y(\omega) = \int_{-\infty}^{\infty} y(t) e^{-j\omega t}\, dt \tag{3.11}$$

The total energy of the impulse y(t) is finite and is represented by the time integral of an instantaneous "power"

$$W = \int_{-\infty}^{\infty} y^2(t) dt \tag{3.12}$$

From Parseval's theorem,

$$\int_{-\infty}^{\infty} y^2(t)\,dt = \frac{1}{2\pi} \int_{-\infty}^{\infty} |Y(\omega)|^2 d\omega \qquad (3.13)$$

The quantity $|Y(\omega)|^2$ is called the energy spectral density; and when plotted against frequency (positive only) gives the energy spectrum. The area under the spectrum curve, multiplied by $1/\pi$, must equal the total energy W. The dimensions of the energy spectral density are energy per cycle per second.

There are three general ways to evaluate the Fourier transform of the impulsive sounds. The analytic approach involves evaluating the integral (3.11); this is possible only when the analysed waveform is represented in the form of an analytical equation. In case of the experimentally recorded signals, it is possible to obtain the solution only for simplified analytical models approximating the signature.

The second approach involves the analog analysis of recorded signatures by means of a set of analog (real) filters. This can be accomplished by recording several repetitions of the analysed pulse on a magnetic tape loop and then analysing the series of repetitions by a wave analyser (which incorporates the filters: Olsen [79]).

As a third approach, the analog filter technique is replaced by digital analysis in computers or by specially designed digital spectral analysers, using the procedure that we have referred to as Fast Fourier transform (FFT).

In general, for the real-valued record y(t) the Fourier transform integral (3.13) may be restricted to the finite interval (0, T) so

$$Y(f, T) = \int_0^T y(t) e^{-j2\pi ft} dt \qquad (3.14)$$

Assuming that y(t) is sampled at "N" equally spaced points a distance "h" apart for arbitrary frequency "f" the discrete version of the equation (3.14) can be shown as

$$Y(f, T) = h \sum_{n=0}^{N-1} y(nh) e^{-j2\pi fnh} \qquad (3.15)$$

$$t = nh, \; n = 0, 1, 2, \ldots N-1, \; T = Nh$$

The discrete values of frequency for the Y(f, T) are:

$$f_k = kf_o = \frac{k}{Nh}$$

$$k = 0, 1, 2, \ldots, N-1$$

At these frequencies the transformed values give the Fourier components defined by

$$Y_k = \frac{Y(f_k, T)}{h} = \sum_{n=0}^{N-1} y(nh)\, e^{-j2\pi n \frac{k}{N}} \qquad (3.16)$$

$$k = 0, 1, 2, \ldots, N-1$$

The results are unique only out to $k = N/2$, the Nyquist cut-off frequency.

The Fast Fourier transform procedures were developed to compute the coefficients of the equation (3.16) with reduced computer time compared to standard methods. Most often used procedures are based on the Cooley-Tukey algorithm ([80], [81], [82]).

Because of the nature of the digital Fourier transform the user should be aware of some limitations and errors introduced to the final result; these have been discussed in Chapter 3.3.1.

In the practical situation only a finite block of data can be processed by the FFT procedure. The size of this block determines the frequency resolution achieved, of step size, $\Delta f = 1/Nh$ (where $Nh = T$, total length of the sample). The number of computed Fourier coefficients when operating on real sampled points is exactly one-half the number of sample points in the analysed block (N). The increase of the sampling rate in time increases the maximum frequency range ($F_{max} = 1/2h$); but for the same frequency resolution the computer memory and time requirements must be increased. All the mentioned factors, i.e., cost, memory size available, maximum frequency range, sampling rate, frequency resolution, etc., have to be carefully chosen to achieve the desired result.

The sampling rate in the time domain is usually chosen according to the required frequency range of the spectrum. The basic rule is that the sampling rate must be at least twice as large as the highest frequency component in the sampled signal. All frequencies beyond that limit, if existing in the signal, should be filtered out because of the danger of being analysed as lower frequency components. This effect is called aliasing.

The necessity of truncating the sampled signal in the time domain gives rise to "leakage" or sidelobe effects in the frequency domain, i.e., spreading of the main lobe of the true power spectral density function and adding an infinite number of small side lobes. Half of these lobes are negative. Several different types of time windows are used instead of the simple "box car" function, with carefully designed characteristics for smoothing the spectral samples and reducing the leakage (see Harris [83]). In the case of transient signals, the windowing is usually made with the simple box car window defined by

$$\begin{aligned} u(t) &= 0 && t < -T \\ &= 1 && -T \le t \le T \\ &= 0 && t > T \end{aligned}$$

All spectrum functions of the test pulses in our experiments were calculated by means of the FFT procedure programmed in digital computers (HP 2100 and IBM 360). Some typical energy spectral density functions are shown in Appendix 6.

4. EXPERIMENTAL TECHNIQUE

4.1 Design of Experiment

4.1.1 General Remarks

The objectives of any psychological experiment are to measure and interpret psychological events. In such experiments the investigator is looking for a relationship, in the simplest case, between two variables, the independent and the dependent variable. The independent variable is manipulated by the experimenter and the dependent one is manipulated by the variations of the independent variable.

The relationships determined by the experiment must be also applicable to the situations other than the one in which the data were obtained; therefore it is important to assure the generality of the experiment. The basis for the generality of the data is representative sampling, which will provide the same composition as for the general population. There are several established procedures to assure representativeness of the sample. They include random sampling with replacement method, random sampling without replacement (in both, each subject from the population has an equal chance of being selected), and rational procedures in which the sample is composed in the same way as the general population. All of these methods, however, have many limitations in practice, and it is almost never possible to obtain truly representative samples from the population.

There are always numerous uncontrolled variables interfering with the investigated relationships in the psychological experiments; for example, sex, age, elements of the environment, etc. Keeping the experimental sample homogeneous in some respects (e.g., same age or sex in the experimental group) may eliminate the extraneous variables, but it must be applied with caution, since it will also influence the generality of data.

A good way of minimizing the effect of uncontrolled variables is randomization. This is, however, most effective for a large number of samples. Nonselective community surveys provide an example of this approach.

4.1.2 Loudness Determination Methods

There are several well established methods for loudness determination available in experimental psychology. Stevens [84] lists seven of them. The most often used in psychoacoustic research are:

(i) Method of adjustment - observer adjusts stimulus until it is subjectively equal to, or bears some desired relation to, the criterion.

(ii) Minimal change procedure - stimulus is varied upward and/or downward and observer communicates its relation to a criterion.

(iii) Paired comparison or constant stimuli procedure - stimuli are presented in pairs. Each stimulus is paired with another or with a reference. Subjects indicate which stimulus in each pair is greater in respect to a given attribute.

(iv) Rating scale or magnitude estimation procedure - each of a set of stimuli is given an "absolute" numerical rating in terms of some attribute.

Each of these techniques has specific advantages and should be used according to the purpose of the investigation. The most suitable for loudness measurements and most often used in loudness research are the magnitude estimation and paired comparison methods. They are frequently combined with the adjustment procedure.

The adjustment procedure gives the subjects control over the stimuli through some sort of "transforming" device. Robinson [85], Stevens and Poulton [86], and Rood [63], among others, used knobs or attenuators as a means of control. A controlling system regulated by a subject introduces a bias which will be dependent upon the characteristics of the device. Stevens and Poulton, for example, reported some differences between results obtained with a sone potentiometer and those with a decibel attenuator. The use of the adjustment procedure, on the other hand, reduces the experimental time and allows more precise measurement of loudness levels.

The method of paired comparisons is especially suitable as a means of establishing subjective equality. It gives the relative levels of two sounds at which they are considered equally loud (or equally annoying). The method is considered more natural than the others, and it usually gives more consistent results. Yet there are some specific problems associated with it. Subjects tend to judge the last presented stimulus in the pair as less acceptable than the first. This can be corrected, however, by alternating the presentation order, and this method was used, for example, by Shepherd and Sutherland [6].

The method of paired comparisons does not give any information about quantitative differences between two compared signals, or deviations from equality with respect to some criterion; thus it is usually suitable only for relative equality level measurements.

The magnitude estimation techniques result from Stevens' numerical scaling procedures [87], [88]. Usually the standard stimulus is presented and observers are given some particular number describing its loudness, noisiness or annoyance, or they may be free to choose their own number. Often there is no standard signal in the experiment and the observers are asked to assign numbers which appear to them appropriate in proportion to a criterion. It has been indicated that the results of magnitude estimation tests may vary with the choice of an intensity of a standard stimulus and its relation to the numerical value assigned to it (Stevens [88] and [86], Hellman and Zwislocki [89] and [90]), and the relative frequency with which the standard is presented, etc.

Attempts to compare the techniques of paired comparisons and magnitude estimation have indicated that both methods provide highly reliable data. While the magnitude estimation technique appears to be the faster, the paired comparison technique seems the more natural for the subjects (Clark and Kryter [91]).

4.1.3 Statistical Considerations

An experiment must be planned so that the results are not biased by irrelevant variables. To assure this, the investigator should introduce randomization into the experimental procedure (e.g., in order of presentation of the experimental stimuli) and use some sort of strictly objective measure of error.

There are well known methods developed in parametric statistics which are widely used for testing the statistical significance of psychological experiments. They allow the evaluation of whether the independent variable has any influence on the dependent variable and often it is done by assuming that there is no such influence. This notion is called the null hypothesis.

A frequently used method for testing the null hypothesis is a "t" test, which is regarded as a most powerful test in statistics. Two general assumptions must be satisfied for usage of the "t" test. The sample from which the data are obtained is assumed to be normally distributed and the variances of the experimental groups are assumed to be equal. Recent studies, however, have indicated that the "t" test remains accurate despite violation of these assumptions, providing they are not large (Anderson [92], Boneau [93], Sheridan [94]).

The "t" test is usually applied to paired samples (it may also be applied to samples which are not paired - Goulden [95]), and it compares the standard deviations of data with the mean difference between samples treated as a variate.

Another powerful tool in statistics used routinely in experimental psychology is an analysis of variance and Fisher "F" ratio test. The analysis of variance is also based on the assumptions that experimental errors are independent in the probability sense, have equal variance, and are normally distributed. The complete analysis of variance sorts out and estimates the variance components, within the groups and between the groups, and in the second place provides for a test of significance. The "F" ratio is the ratio between "between groups" variance and "within groups" variance, and allows one to test the null hypothesis in case of more than two samples.

4.2 Experimental Procedure

Our investigative work consisted of four experimental test series involving human observers. Each series was designed to test the subjective loudness of a different type of transient sound; i.e., sonic-boom N-wave signatures, minimized "low-boom" signatures, filtered N-wave signatures and idealized "quarry blast" signatures.

All subjective tests were carried on in the UTIAS Sonic Boom Simulator (for more details, see Chapter 3), and the paired comparison technique was used. All test signals series were generated by computer and then recorded on FM tape recorder; thus the subjects taking part in the comparison tests had no control over the compared signals.

During the 15-20 minute test sessions the signals in pairs were presented to each subject while seated singly in the simulator booth. They were required to identify which sound in a pair was judged to be the louder, and to communicate this verbally, through the intercom, to the experimenter.

Three judgement scores were used: "louder", "may be louder" and "equal loudness" (similarly for annoyance when annoyance was a criterion).

Thus a set of five numerical scores was obtained:

Test pulse louder	= -2
Test pulse may be louder	= -1
Both equally loud	= 0
Reference pulse may be louder	= 1
Reference pulse louder	= 2

The reference and test signals in pairs were presented twice to the subjects, each time in the opposite order; i.e., AB and BA. The test series for each compared pair of signatures consisted of three "warm up" pairs at the beginning of the series, and several comparison pairs, each with different overpressure ratios between the test and reference signals. They were presented to the observers in random order.

The test series were generated according to the recommendations for subjective comparison tests formulated by Reichardt and Niese [96]. The intervals between the compared pulses were kept between 500 and 1000 milliseconds assuring no mutual interference and, on the other hand, small possible memory errors. The duration of the comparison pulses never exceeded 2:1 ratio and the continuous sound in the last part of the experimental project was presented for 400 ms.

The individual results for each value of the tested parameters (e.g., rise time or duration) were plotted in the form of graphs - relative loudness (in scores) versus overpressure ratio between the test and the reference signals. Typical examples of such graphs are shown in Fig. 15. From each graph the overpressure ratio for equal loudness (score = 0) was determined and the average of these values over all populations of participating subjects was used to construct the final equal loudness curves.

UTIAS male graduate students were used as subjects in all experiments. Audiograms were obtained before and after each experimental session for all observers. In addition they were examined prior to the experiment by a qualified otolaryngologist and found to have healthy ears and hearing levels no more than 15 dB down from the ISO audiometric zero [27], except for occasional dips to -20 dB in the frequency range above 4 kHz. This is well within the range of hearing (0 to 25 dB down) considered normal. Before each test series careful instructions (see Appendix 5) were given to the individual subjects.

All the empirical results were compared with the theoretical calculations based on the Johnson-Robinson prediction procedure [2] developed for sonic-boom signatures.

The analysis of variance, "F" Fisher ratio test, and "t" test were used as a standard tool for testing statistical significance of the experimental results.

5. SUBJECTIVE EXPERIMENTS

5.1 Subjective Loudness of N-Wave Sonic Boom

5.1.1 Introduction

There is continuing interest in human response to sonic-boom type pressure waveforms. In particular, the role of the rise time and duration of the N-wave signatures in controlling subjective loudness has been under study.

The typical values of sonic-boom characteristic parameters measured at ground level depend upon the type of aircraft, characteristics of the flight and atmospheric conditions. In general, the peak pressure Δp varies from 1 to 10 psf and duration varies from 0.05 to 0.3 of a second (NASA measurements [75]). Typical values for Concorde SST aircraft are 1 to 2 psf overpressure and from 200 to 250 ms duration. Reported rise time values for Concorde range from 0.1 to 15 ms [97]. Measurements for various military aircraft made, for example, during the experiments in Edwards AFB appear to show rise time in a range from 0.2 to 45 ms. True rise times may be even shorter, as 0.1 to 0.2 ms represent instrumental rise time limits.

Experimental subjective studies with sonic-boom signatures in laboratory environments have been conducted by Zepler and Harel [1], Lukas and Kryter [60], Kryter [65], Shepherd and Sutherland [6], and Rood [63].

The theoretical prediction procedure for determining the apparent loudness of sonic-booms has been proposed by Johnson and Robinson [2] following earlier proposals by von Port [39], and Zepler and Harel [1] who developed their own calculation procedure for determining the loudness of sonic-boom.

The empirical studies indicated significant change in loudness of N-waves with variation of rise time and marginal change in loudness with variation of duration. Theoretical predictions based on the spectrum of sonic-boom signatures show a similar relationship.

Shepherd and Sutherland, as well as Rood, have performed tests at different reference overpressure levels. Their results indicate that the level appears to have little effect on the relative effects of rise time and duration in the range from 0.8 to 2.4 psf in Shepherd's experiments and between 0.5 and 2.0 psf in Rood's tests.

This segment of the present investigation is very similar in concept to the investigation of Shepherd and Sutherland. However, advances in the present simulation booth design allowed for a five-fold shorter rise time.

5.1.2 Results and Discussion

Two separate sequences of sonic-boom comparisons featuring N-wave signatures were carried out with twenty subjects. Examples of N-wave booms reproduced in the Sonic Boom Simulation Booth and used in the current experiment are shown in Figs. 16 and 17).

In the first sequence of experiments the boom duration "D" was held constant at 200 ms, the rise time "τ" was varied over the range 0.22 to 10 ms, and the peak overpressure over the range 0.5 to 4 psf (24 to 192 N/m^2) the latter only for the longer rise times. For each rise time the overpressure of the test

N-wave was adjusted so that the observer could judge for which overpressure level the loudness matched that of a reference N-wave with 1 ms rise time, 1 psf (48 N/m^2) overpressure, and 200 ms duration. In this fashion contours of equal loudness versus rise time were developed.

In the second sequence of experiments the test sonic-boom rise time was held constant at 1 ms and its duration D was varied from 100 to 250 ms; a second equal loudness contour (overpressure ratio vs duration) was defined by comparison tests, adjusting the overpressures from 0.5 to 2 psf (24 to 96 N/m^2). The reference N-wave was the same as the previous one.

The two experimentally determined equal loudness contours for the N-wave signatures are plotted in Figs. 18 and 19. The overpressure ratio is defined by $\Delta p_{\text{level ratio}}$, where

$$\Delta p_{\text{level ratio}} = -20 \log_{10} \frac{p_{\text{test}}}{p_{\text{ref}}} = -20 \log_{10} \frac{p_{\text{test}}}{1 \text{ psf}} \qquad (5.1)$$

Each subject carried out approximately 180 judgements during the course of two test sequences. The experimentally determined standard deviation for each plotted point is indicated by the vertical bars on the graphs. It was noted that the deviations among the individual comparison results increased progressively as the differences between the features of the reference-boom signature and the test-boom signature increased. This reflects the increased comparison difficulties. The standard deviation is typically about 1 dB; but for booms having a duration of 250 ms (rise time 1 ms) it rises up to 1.4 dB and for booms having rise time of 10 ms (duration 200 ms) it is about 3.3 dB.

The one-way analysis of variances performed on the experimental data along with the individual results are shown in Appendix 1. Each table (i.e., Tables 1 and 4) represents averaged individual results over the AB and BA orders of presentation. At the bottom are shown the calculated mean values for all subjects.

The analysis of variance for the rise time tests yields a large sum of squares representing variability among the population means (SSB), but much smaller values of the component representing variability within the samples (SSE), i.e., between the subjects.

The Fisher's "F" ratio is significant at the significance level 0.01. Therefore the null hypothesis can be rejected, and we conclude that the experimental differences in the rise time comparison are real and not due to chance.

In the case of the duration comparison the Fisher "F" ratio for data obtained in the experiment, excluding the sequence with 250 ms duration, is much smaller than the value required for rejecting the null hypothesis. Therefore the null hypothesis cannot be rejected, and we conclude that there are no real differences between loudness of the sonic booms over the duration range 100 to 200 ms. The same Fisher ratio calculated over all points, including signatures with 250 ms duration, indicates significant difference between mean values at the confidence level of 0.01.

Along with the present results (labelled Niedzwiecki), Fig. 18 reproduces the experimental results of Shepherd and Sutherland [6] along with the calculated theoretical predictions by Zepler and Harel [1] and by Johnson and Robinson [2]. There is generally good agreement over the common range, essentially within the error bars. The predicted decrease in loudness with increasing rise time is very marked above 0.5 ms. At 10 ms rise time the results are somewhat divergent, but with a large experimental uncertainty.

In the rise time range between 1 ms and 10 ms the equal loudness curve falls 8 dB which compares with the value of 13 dB in the Shepherd and Sutherland tests and 9 dB predicted by Johnson and Robinson's theory. In the range below 1 ms the equal loudness curve rises an additional 3 dB at the lowest rise time of 0.22 ms investigated in the present test.

In the range of durations from 100 ms to 200 ms the loudness appears to be independent of duration changes; however, for durations of about 250 ms the experimental data indicate an abrupt rise of the equal loudness curve. Both Shepherd and Sutherland (experiment) and Johnson and Robinson (theory) find a negligible influence of the duration on the subjective loudness for the total experimental range.

5.1.3 Conclusions

The present results appear reasonably consistent with earlier theoretical and experimental subjective boom data, except for the effects of the longer boom durations (in excess of 250 ms) shown in Fig. 19. The substantial rise in the equal loudness contour in this case remains unexplained; however we may speculate that this effect may be due to excitations of body vibration by the boom. Dempsey and Leatherwood [98] have shown experimentally that the human body sensitivity for vibrations (with discomfort as a criterion) is maximal in the frequency range from 3 to 8 Hz. The dominant frequencies of sonic boom, proportional to reciprocal duration, lie in this range.

The essentially good agreement with previous data adds confidence to our experimental technique as well as to the existing theoretical methods for predicting the subjective loudness of N-wave signatures (especially the Johnson and Robinson procedure). Compared with earlier experience these are now seen to validate over an expanded parameter range given by

Rise time	0.22 to 10 ms
Duration	100 to 250 ms

5.2 Subjective Loudness of "Minimized" Sonic Boom Waveforms

5.2.1 Introduction

One of the major problems that has limited development of supersonic civil aviation is annoyance caused by sonic booms generated by overflights. Therefore, a prominent avenue of research has been the exploration of techniques for minimizing or possibly altering the characteristic N-wave produced in the far-field by present day supersonic aircraft.

McLean [7] observed that for sufficiently long aircraft (e.g., ≥ 300 ft (~ 90 m)) the "midfield" sonic-boom signature may not have evolved fully into an N-wave at ground level. He suggested that the shape of the aircraft cross-section and lift distribution could be modified to optimize this midfield waveform for reduced subjective loudness.

Hayes [99] pointed out that in the real atmosphere, because the characteristics coalesce more slowly than in the uniform atmosphere, the "midfield" signature "freezes" instead of evolving into an N-wave at ground level. "If midfield effects persist to $\pi H/2$ real-atmospheric scale heights (H) in homogeneous atmosphere, they will persist indefinitely below the aircraft in real atmosphere." Tailoring of the midfield-type sonic-boom signature for minimum boom was developed and extended by George and Seebass in the series of papers, e.g., [8], for flight in an isothermal atmosphere. They exploited the phenomenon of "freezing" of the midfield signatures.

The mathematical theory of Seebass and George has been extended to apply in detail to the real atmosphere by Darden [9], [10]. Her formalism permits minimization of either the initial shock of the signature or maximum overpressure, compared with an N-wave. Again this is accomplished by means of an especially tailored distribution of the aircraft cross-section and lift.

By means of such tailoring Darden computed a family of minimized or "low-boom" signatures associated with certain proposed "second generation" supersonic transport configuration (Fig. 20) The expectation was that for given aircraft volume, weight, flight altitude and Mach number, these signatures should sound less loud than normal N-waves. Our objective in the present test series has been to test this notion experimentally. The signatures would be simulated in the UTIAS Sonic Boom Simulator, and jury tests of the subjective loudness would be conducted. Finally the results would be compared with calculated values by the Johnson and Robinson method.

Darden's signatures are not quite symmetric fore and aft (cf. Fig. 20). However, the Johnson-Robinson method for predicting loudness is predicated on fore-aft symmetry. For this and other reasons of a practical nature, Darden's signatures were replaced by symmetric ones in the tests, the relationship being as in Fig. 20. The differences are not great and it is thought their effect on the subjective loudness of these "low-boom" signatures should be minimal.

The following series of subjective tests attempt to establish empirically the relationships between the subjective loudness and "low-boom" signature characteristic parameters, i.e., the flat top duration D_1, and the ratio shock overpressure/peak overpressure ($x = \Delta p_{SH}/\Delta p_{MAX}$; cf. Fig. 20).

5.2.2 Results and Discussion

The "low boom" or "flat top" sonic boom signatures did not reproduce accurately in the UTIAS simulation booth, despite the equalization filter adjustments. This was resolved by means of the computer-aided scheme for "predistorting" the input signal (see Chapter 3.3 for details). The resulting typical pressure signatures recorded inside the simulation booth as used in the subjective tests are shown in Fig. 21.

In all test sequences the signatures had a fixed rise time (1 ms) and total duration (150 ms).

Two separate comparison sessions were carried out. In the first one the flat top duration of the signatures was also held constant (D_1 = 30 ms) and the ratio $x = \Delta p_{SH}/\Delta p_{MAX}$ (front shock overpressure/maximum overpressure ratio) was varied within the range 0.2 to 1.0. The equal loudness contour (overpressure ratio of test and reference signals versus "x") was defined through the comparisons of these signatures with the reference N-wave signature having the same rise time (τ = 1 ms) and total duration (D = 150 ms), and overpressure of $\Delta p_N = 0.5$ psf (24 N/m^2). Ten observers, all UTIAS male graduate students, took part in this experiment; each of them carried out about 120 judgements.

In the second test sequence the overpressure ratio $x = \Delta p_{SH}/\Delta p_{MAX}$ was held constant at the level x = 0.5 and the equal-loudness contour of overpressure ratio vs flat top duration (D_1) was determined for the "low-boom" signatures having a flat top duration within the range 10 to 60 ms at a total duration of 150 ms (i.e., from 0.0667 to 0.4 of the total duration). The reference N-wave had the same total duration (150 ms) and rise time (1 ms) as previously, but the overpressure was fixed at 1 psf (48 N/m^2). Eight observers took part in this experiment, and each carried out about 100 judgements during the test sequence.

Two equal-loudness contours derived from the experimental results for the "low-boom" signatures are illustrated in Figs. 22 and 23. The first shows the overpressure level ratio vs $x = \Delta p_{SH}/\Delta p_{MAX}$, and the second shows the overpressure level ratio vs flat top duration D_1. The overpressure level ratio is defined by

$$\Delta p_{\text{level ratio}} = -20 \log_{10} \frac{\Delta p_{MAX}}{\Delta p_N} \qquad (5.2)$$

where Δp_N = overpressure of the reference N-wave,

Δp_{MAX} = maximum overpressure of the test "low-boom" signature.

The plotted equal-loudness curves are based on the averaged values calculated from the experimental results for each subject. The vertical bars indicate the experimentally determined standard deviation. The standard deviations for the ratio "x" comparisons are within the range 0.8 to 1.5 dB; for the flat top duration D_1 comparison they range from 0.6 to 1.4 dB. The detailed results for each experimental series and all individual results are shown in Appendix 2, along with the result of the one-way analysis of variance.

The Fisher ratio "F" is significant for variable "x" comparisons at the significance level 0.01; thus the differences between the means for the different values of ratio "x" are taken to be real.

The analysis of variance performed on the results of the second test series (with varying flat top duration D_1) does not permit rejection of the null hypothesis; thus the differences between mean values of the subjective loudness for different flat top durations are taken to be the result of chance.

It was found that to maintain equal loudness the overpressure ratio must be increased by 11.7 dB as the parameter "x" increases from 0.2 to 1.0 (Fig. 22). The actual properly scaled waveforms judged as equally loud are shown in Fig. 24. The comparison suggests that subjective loudness of the "low-boom" type of signature depends mainly upon the front (and rear) shocks. This conclusion is considered valid as long as the ratio τ/D_1 is of the same order as used in the experiment (i.e., $\tau/D_1 = 0.033$), or smaller.

The results (for the particular value of ratio τ/D_1, rise time and duration used in the test) are well approximated by the formula:

$$\Delta p_N \text{ (N-wave)} = \Delta p_{SH} + 0.11\ \Delta p_{MAX} \text{ ("low-boom")} \tag{5.3}$$

for equal loudness, provided the N-wave has the same rise time and duration (Fig. 25). This tells us that the peak pressure Δp_{MAX} contributes only one ninth as much to the loudness as the front shock (and similarly for the rear shock); that is, the front (and rear) shock amplitudes (for fixed rise time) do dominate the loudness. The influence of the Δp_{MAX} on the subjective loudness will be higher for higher values of ratio τ/D_1.

For the effect of the "flat top" duration on the subjective loudness refer back to Fig. 23. The overpressure level ratio for equal loudness varies less than 1 dB with increase in the flat top duration from 0.0667 to 0.4 of the total duration. This change is within the range of the experimental error and was not found statistically significant. Therefore, we can infer that the duration of the flat top part of the "low-boom" signature has a negligible influence on the subjective loudness in the experimental range of values (similarly for the ratio τ/D_1).

The results of the "low-boom" comparison tests were supported by the theoretical loudness calculations. The loudness of each signature judged equally as loud as the reference N-wave was calculated from the energy spectrum density function obtained through the FFT procedure in a digital computer. A few examples of the energy spectrum for the "low-boom" signatures used in the experiment are shown in Appendix 7.

The Johnson-Robinson method for N-wave sonic-booms, in a version based on the Stevens Mark VI procedure for continuous sounds, was followed in the calculations. The loudness was calculated for the positive parts of the signatures only, which is justified after Johnson and Robinson on the ground that the separation between front and rear shocks is sufficiently long compared to the critical time of the auditory system.

The results of these calculations are compared with the calculated loudness of the reference N-waves in Figs. 26 and 27. The calculated loudness (in phons) for all studied "low-boom" signatures differs from the calculated loudness of the reference N-waves which sound equally loud by less than 1 phon. This very good agreement of the empirical and theoretical results, in terms of the relative loudness, supports the viability of the Johnson-Robinson loudness comparisons between N-wave and the "low-boom" family of signatures within the range of parameters given by:

$$\left.\begin{aligned} 0.0667 &\le (D_1/D = D_2/D) \le 0.4 \\ 0.2 &\le (x = \Delta p_{SH}/\Delta p_{MAX}) \le 1.0 \end{aligned}\right\} \text{ at } D = 150 \text{ ms}, \tau = 1 \text{ ms}$$

5.2.3 Conclusions

A series of jury tests of the perceived loudness of the "low-boom" sonic-boom signatures has been completed and the results compared with theoretical predictions. The comparisons indicate that the loudness of these signatures is dominated by the amplitude Δp_{SH} of the front (and rear) shocks. The peak amplitude can thus be much larger than that of an N-wave that sounds equally loud (assuming that the ratio τ/D_1 is of the same order as in the experiment or less). Put another way, an N-wave of the same peak amplitude will sound much louder than the "low-boom" signatures with small ratio "x". Based on Darden's calculations [9], [10] of possible "low-boom" signatures for realizable aircraft, with $\Delta p_{SH}/\Delta p_{MAX}$ as low as one half, the attainable loudness reductions are roughly equivalent to those resulting from halving the present N-wave amplitudes.

There is an important caveat concerning the above results: they refer solely to sonic booms as heard outdoors. Indoor sonic booms are quite different: their waveforms as well as amplitudes have been grossly modified by the transmission characteristics of walls and windows. These are normally dominated by low frequency resonances, and are quite insensitive to the shock amplitudes and rise times. The major parameters governing indoor boom intensity are the impulse (area under positive half of outdoor waveform) and duration. For the "low boom" signatures these parameters are not much altered compared with the standard N-wave: indeed, the impulse may be increased. Thus the "low boom" signatures, although much quieter outdoors than N-waves for comparable size aircraft, offer no advantage indoors.

The relative loudness predictions of the Johnson-Robinson theory conformed very well to the measurements. Thus their potential for applicability to a much broader range of transient sounds is indicated.

The above results make it clear that the rise time of the shock waves in impulsive sounds is a major parameter controlling the subjective loudness, along with their peak amplitude.

5.3 Subjective Loudness of Filtered N-Wave Signature

5.3.1 Introduction

Analysis of the energy spectral density of a typical N-wave signature reveals that there are two specific break points in the gross envelope of the spectrum function (see Fig. 1, after Johnson and Robinson [2]). The envelope, starting from the low frequency end of the spectrum, rises at 6 dB per octave up to a peak frequency of $\sqrt{3}/\pi D$ Hz (D = duration), then falls at 6 dB per octave and finally from a frequency of $1/\pi\tau$ Hz (τ = rise time) drops at 12 dB per octave. Typically, the first maximum occurs in the region ranging from 2 Hz to 6 Hz for current supersonic aircraft (i.e., duration ranges from 100 to 300 ms). Thus most of the energy of the sonic-boom N-wave signature is concentrated far below the normal audio frequency range of the human ear. We know from well established evidence that the sensitivity of the ear is extremely low at frequencies below the range of 20 Hz, called "infrasonic". Whittle et al [28] have shown that the decrease of the equal-loudness curves in the range from 25 to 3 Hz is approximately 60 dB at the levels from threshold up to 70 phons.

Experimental laboratory investigations of human and animal response to sonic-boom are usually carried out in specially designed loudspeaker facilities in order to fully reproduce N-wave signatures, including the infrasonic content. Thus frequency-wise, it is necessary to extend the flat response of such a simulation facility down to nearly DC. This can be done by defeating the usual low-frequency roll-off by use of either an airtight chamber or earphones with an effective pressure seal (see Chapter 3). The question arises: What influence upon the subjective human response has the intense infrasonic part of the sonic-boom spectrum in the situation of whole body immersion in the acoustic field? If this influence were found to be small or negligible, it would be possible to avoid building the costly simulating equipment; instead the subjective sonic-boom tests could be conducted with relatively inexpensive high fidelity audio equipment with frequency response from 20 to 10,000 Hz.

An indication that the energy below approximately 40 Hz has an insignificant effect upon the loudness was given by Zepler and Harel [1] in 1965. They used an analog (real) high pass filter with cut-off frequencies ranging from 20 to 240 Hz and reproduced the filtered N-wave signatures with specially designed earphones. Introduction of high pass filters with cut-off frequency of either 20 or 40 Hz was found to have no effect on the subjective loudness of sonic booms of rise times of 3 and 1 ms. The results of Zepler and Harel agree with later theoretical estimations by Rood [63]. He calculated the correlation between the energy in discrete bands of the sonic boom spectrum and the subjective loudness level of the stimuli. Correlations in the frequency bands of 5-10 Hz, 10-20 Hz and 20-40 Hz were found to be insignificant; the 40-80 Hz band was found significant at the 5% level, and the remainder of the level below 1%. The range of parameters used in this comparison were: rise time varying from 2 to 16 ms and peak pressure varying from 0.5 to 2 psf.

The present test series is essentially a repetition and refinement of the Zepler-Harel experiment [1] in the situation of whole-body immersion in the acoustic field. The analog high pass filters are replaced by digital filtering, and the filtered waveform is further improved by the "predistortion" technique. The results on subjective loudness are compared with the predictions of the Johnson-Robinson theory [2] to provide a further test of that theory.

5.3.2 Results and Discussion

Fifteen male subjects were tested with loudness as a judgement criterion and ten with annoyance as a judgement criterion. All of them had been checked, as described earlier, for normal hearing, and all were UTIAS graduate students. Each subject carried out about 40 judgements.

The paired comparison technique, as in the previous experiments, was used. The reference N-wave with 1 psf (48 N/m^2) overpressure, 1 ms rise time and 150 ms duration was generated in "predistorted" form by a computer. Two test signatures were obtained from the predistorted reference signature by means of a digital high pass filter having two cut-off frequencies: 25 and 50 Hz. The details of the filtering process have been given in Chapter 3.3. Typical pressure signatures recorded inside the booth are shown in Fig. 14.

In the first test series the overpressure ratio levels defined as

$$\Delta p_{\text{level ratio}} = 20 \log_{10} \frac{\Delta p_F}{\Delta p_{ref}} = 20 \log_{10} \frac{\Delta p_F}{1 \text{ psf}} \qquad (5.4)$$

were found for the conditions of equal loudness for the respective filtered signatures compared with the reference N-wave. Based on the values of the respective signature overpressures judged as equally loud, the absolute subjective loudness levels for these signatures were calculated using the Johnson-Robinson [2] procedure. Both experimental and theoretical results are shown in Table 3.1. The first row contains the empirically established overpressures of the signatures judged by the observers as equally loud (values shown represent the averaged results). Comparison indicates that the loss in infrasonic frequency content due to filtering must be compensated by an increase in overpressure for equal loudness. The differences, which increase with cut-off frequency, are however very small.

Table 3.1

Results of Filtered Sonic-Boom Comparison Experiment

	Reference Rise Time = 1 msec Duration = 150 msec	Filtered $f_{cut\text{-}off}$ = 25 Hz	Filtered $f_{cut\text{-}off}$ = 50 Hz
Overpressure for equal loudness	Δp = 1 psf	Δp = 1.01 psf (+0.09 dB)	Δp = 1.05 psf (+0.42 dB)
Calculated loudness	105 phon	104.9 phon	104.4 phon
Overpressure for equal annoyance	Δp = 1 psf	Δp = 0.94+psf (-0.52 dB)	Δp = 0.94-psf (-0.55 dB)

Standard deviations calculated from the test data were 0.88 dB for a cut-off frequency of 25 Hz and 1.36 dB for a cut-off frequency of 50 Hz. This reflects growing difficulties in loudness comparison for higher cut-off frequencies because of the increasing differences in character between compared sounds.

The individual results along with statistical analysis are shown in Appendix 3. The "t" test applied to the results of the loudness comparisons show no significant difference between mean values obtained, indicating no real difference in loudness between filtered N-wave signatures for the two cut-off frequencies of the high pass filter: 25 and 50 Hz.

The theoretically calculated loudness levels of the reference sound and test signatures (second row in Table 3.1), which were found to be equally loud through the subjective test, differ by less than 0.6 phon. This good agreement provides further support for the viability of the Johnson-Robinson loudness prediction procedure, especially their special method for the evaluation of the low-frequency bands (see Chapter 2.4.3).

All subjects during the loudness comparison session were asked about the difference in character between the test and reference signals in terms other than loudness parameters. All remarked on relatively easily detected differences in "sharpness", frequency content, and subjective duration. This led us to the conclusion that, even though the subjective loudness of sonic boom does not depend upon the energy below 50 Hz range, other subjective parameters may be affected by this part of the spectrum.

Another test sequence similar to the one described above was carried for ten subjects with annoyance as a criterion of judgement. The overpressures for equal annoyance, averaged over all observers (third row in Table 3.1), show an opposite tendency from the case of loudness comparison. To state it simply, annoyance was slightly increased as a result of the filtering out of the low frequency sound energy.

A similar phenomenon has been reported by Turner and Burns [100] who found that subjects comparing a 5 second long burst of one-third octave band white noise centred at 3150 Hz and the same burst with additional one-third octave band white noise centred at the much lower frequency of 250 or 500 Hz judged the first one as the more annoying.

The standard deviation for the annoyance comparison test was for both cases equal to 1.0 dB.

The individual results are shown also in Appendix 3. In the same appendix are shown the results of the statistical analysis performed on the experimental data. The "t" test indicates no significant differences between means at cut-off frequency of 25 and 50 Hz for annoyance as a judgement criterion. The same "t" test performed over the results obtained in the tests with different criteria (i.e., loudness and annoyance) but the same cut-off frequency does not allow rejection of the null hypothesis for the cut-off frequency of 25 Hz, but gives the t ratio value close to the value required for the 0.1 level of significance. For the cut-off frequency of 50 Hz the calculated value of "t" allows rejection of the null hypothesis at the 0.1 level of significance and is very close in value to the "t" parameter required at the 0.05 level of significance.

Therefore, the differences between the results of the subjective experiments with loudness and annoyance as judgement criterion are accepted as real and due to factors other than chance at a 0.1 level of significance, at a cut-off frequency of 50 Hz.

5.3.3 Conclusions

A series of test comparisons between unfiltered and high-pass filtered N-wave signatures (cut-off frequencies of 25 and 50 Hz) was carried out. The loudness level of the filtered signatures, with their reduced low frequency content, was found to be only marginally lower than that of the comparable unfiltered N-wave (both signatures had a rise time of 1 ms, duration of 150 ms). This result was consistent with the theoretical loudness predictions based on the Johnson-Robinson [2] method. It was also found compatible with the previous results obtained by Zepler and Harel [1] (experimental) and Rood [63] (calculated).

The above result conformed very well to one of the general conclusions of the N-wave investigation in Chapter 5.1: there was no significant change in subjective loudness with variation of the N-wave duration (except for the anomalous abrupt rise in excess of 250 ms). Increased duration increases the energy in the infrasonic frequency range: thus insensitivity to duration translates into an insensitivity to the infrasonic frequencies in the N-wave.

The second test sequence with annoyance as a criterion indicated a slightly increasing tendency in annoyance with reduction of the low frequency energy in the N-wave signature. A similar phenomenon was reported by Turner and Burns [100].

The difference between loudness and annoyance of the filtered signatures were found statistically significant at 0.1 level of significance for the case of high pass filtering with cut-off frequency of 50 Hz.

5.4 Subjective Loudness of Idealized "Blast" Signatures

5.4.1 Introduction

In three preceding paragraphs (i.e., 5.1, 5.2 and 5.3) it has been shown that the relative loudness predictions based on the Johnson and Robinson [2] theory, agree very well with subjective loudness measurements for three types of transient sounds: the N-wave signature, the "low boom" signature and filtered N-wave signature.

The last part of this project was designed to extend the applicability of this theory to even broader types of transient sounds and also to verify the absolute values of the theoretically calculated loudness levels by the Johnson and Robinson method.

Considerable attention has been devoted recently to research on quarry blasts' effects on human population, by various government agencies, due to the numerous complaints. For this reason the impulsive sounds chosen for subjective testing were based on the quarry blasts, measured for example by Taylor et al [11]. The range of the blast signature parameters recorded in Taylor's project extends for peak overpressure from 0.42 psf (20.2 N/m^2) to 4 psf (192 N/m^2) and for duration of the signature from 0.14 to 2.7 seconds. An example of pressure-time history given in this report is shown in Fig. 28.

The pressure signatures of a series of quarry blasts show great individual variations. Figure 28 is, in fact, more regular in its features than most. Thus it has been thought worthwhile to suppress the variation in signature shape by idealizing the waveform. The idealized signature, shown in Fig. 29, resembles the real signature of Fig. 28 in having a series of shocks or echoes (so that it looks like a comb); however, the "teeth" of the comb are all of identical shape and spacing. More precisely, the electrical input signatures, generated in the computer without "predistortion", consist of a different number (M) of successive pulses each 25 ms long with shock-like rise time of 0.22 ms and quasi-exponentially decaying "tail", according to the formula:

$$V = (1 - t/D)\, e^{-at} \tag{5.5}$$

where D = duration of the individual pulse (25 ms),

a = constant ($0.1\ \mathrm{ms}^{-1}$).

The input signatures together with their spectral density functions are shown in Appendix 7, and the pressure waveforms recorded within the booth, and used for the subjective tests, are shown in Fig. 29.

5.4.2 Results and Discussion

In the series of subjective comparison tests (using techniques described in Chapters 3 and 4) the rise time of each individual pulse was held constant at 0.22 ms and the duration of the pulse sequence was varied from 25 to 400 ms; this was done in five steps, the five different signatures being sequences of 1, 2, 4, 8 and 16 individual pulses, respectively. The peak overpressure was varied over the range 0.35 psf (16.8 N/m^2) to 1.5 psf (72 N/m^2).

The reference pulse in each comparison series was always another blast signature, being twice as long as the test one. Thus four comparison tests were carried out and the duration (T) of the signals in pairs were:

25 ms vs 50 ms

50 ms vs 100 ms

100 ms vs 200 ms

200 ms vs 400 ms

This arrangement allowed us to keep the duration ratio between compared signals within the recommended value 2:1 (see Reichard and Niese [96]), since a much larger disparity in duration makes loudness comparisons difficult. Furthermore, this procedure permitted covering a wide range of durations.

In this fashion contours of equal loudness versus duration (or number of individual pulses in the signature) were developed. Fifteen subjects took part in the series of tests, each making about 90 comparisons. The individual results along with their statistical analysis are shown in Appendix 4. The results as shown again represent the overpressure ratio of reference and test signature for equal loudness, averaged over the AB and BA order of presentation.

The results averaged over the total population of the observers were then transformed into the ratios between each tested pulse and the new reference level - being the overpressure level of the longest ("continuous") signature, with duration of 400 ms. The overpressure ratio is defined by $\Delta p_{\text{level ratio}}$, where

$$\Delta p_{\text{level ratio}} = -20 \log_{10} \frac{\Delta p_M}{\Delta p_{16}} \qquad M = 1, 2, 4, 8 \qquad (5.6)$$

The resulting experimentally determined equal loudness curve (overpressure level ratio vs duration) is plotted in Fig. 30 re 400 ms case taken as 110 phons (solid line).

The standard deviation varies from 0.75 dB for comparisons between pulse sequences of 200 ms and 400 ms duration, to 1.45 dB for comparisons between pulse sequences of 100 ms and 200 ms duration.

The Fisher "F" ratio and analysis of variance performed on the data (shown in Appendix 4) allows us to reject the null hypothesis at 0.01 level of significance and to conclude that the differences between mean values obtained from the duration comparison are real.

The overpressure level ratio versus duration contour rises about 2 dB for each doubling of duration in the range 25 to 100 ms. For signatures longer than 100 ms the increase is progressively slower, approaching as an asymptote the level for continuous sound (in excess of 400 ms). This type of relationship between the loudness and the duration of the impulsive sounds has been reported previously in the literature and reflects the temporal integration effect of the auditory system (Munson [45], Garner and Miller [40], Green et al [41], Harris [42], Plomp and Bouman [43], etc). It was discussed in Chapter 2.3.2.

As mentioned before (see Chapter 2.3.2), there is little agreement in the literature as to the value of the time constant of the human auditory system. The estimates range from 20 to 200 ms and there are conflicting reports whether the temporal integration function depends upon the spectrum of the stimuli and its intensity level. It is rather hard to accurately measure the critical time of the ear for the averaged data obtained in the current test, because of the relatively large intervals between the experimental values of duration; however we may estimate it as being approximately between 100 and 200 ms.

The observation that the critical time for the test "blast" pulses, estimated from the empirical equal-loudness curve, is longer than 100 ms has a specific consequence for the theoretical prediction of the absolute loudness levels based on the Johnson-Robinson method. They assumed [2] that the time necessary to evoke full response of the human auditory system is 70 ms - the value estimated by von Port [39]. The underestimation of the real value of the critical time, as used in calculations, should lead to a significant overestimation of the absolute values of loudness levels calculated by the Johnson-Robinson scheme. Indeed, the calculated loudness levels for the test "blast" signatures judged as equally loud, using time constant value equal to 70 ms are overestimated by approximately 4 phons compared to the experimental curve (Fig. 30 - curve 3), in terms of the <u>absolute</u> loudness levels. However, in terms of the relative values, the predicted levels are in very good agreement with the data acquired in the experiment up to $\tau_c = 70$ ms: this can be seen by downshifting the 70 ms curve by 3.6 phons, which superimposes it on the experimental curve. The absolute loudness scale in phons on the graph is set up by the calculated loudness level of the signature (using Stevens' Mark VI procedure) with the duration of 400 ms, which can be regarded as a "continuous sound" (see, for example, Reichard and Niese [96]) for human auditory system response. (The term "effective loudness" is used on the graphs as a reminder that the experimental curve is still in terms of the Δp level ratio for equal loudness, as defined by equation 5.6.)

A similar result (i.e., overestimation of the absolute subjective loudness by the original Johnson-Robinson theory) was reported by Rood [63], who compared the experimental measurements of the absolute values of subjective loudness of half N-wave signatures with predictions by the Johnson-Robinson theory. The graph showing this comparison is reproduced in Fig. 31. The theoretical calculations show constant overestimation of the loudness by about 3.2 dB, which is very close to the result obtained in this investigation.

Agreement in both absolute and relative loudness levels between the experimental data and the Johnson-Robinson predictions is attained when we change the critical time value in the calculations to 175 ms (this lies in the range estimated previously from our experimental data); cf. Fig. 30 again, curve 2.

The Johnson-Robinson principle of linear integration of sound energy with time applies only for signal duration $T \leq \tau_c$. For $T > \tau_c$ it is implicit that the judged loudness is constant at the value for continuous sound of the same spectral content. This accounts for the loudness "ceiling" of 110 phons shown in Fig. 30 for the Johnson-Robinson calculation (curve 2). The linear integration assumption, it is seen, starts to break down when the duration exceeds two thirds of the critical time τ_c.

By contrast the experimental loudness curve (solid line in Fig. 30) increases smoothly with duration toward the continuous sound asymptote. It is as though there were some sort of saturation effect in the neurological response. Without attempting any specific modelling inferred from observed cochlear and neurological mechanisms, one can postulate a simple exponential approach to saturation. Thus, in place of Johnson and Robinson's [2] linear law which defines:

$$\text{band pressure level} = \frac{1}{\pi}\int_{\omega_1}^{\omega_2} \frac{|F(\omega)|^2}{T}\, d\omega \times \left\{ \begin{matrix} T/\tau_c & T \leq \tau_c \\ 1 & T \geq \tau_c \end{matrix} \right\} \qquad (5.7)$$

we introduce the generalized definition

$$\text{band pressure level} = \frac{1}{\pi}\int_{\omega_1}^{\omega_2} \frac{|F(\omega)|^2}{T}\, d\omega \left[1 - e^{-T/\tau_c} \right] \qquad (5.8)$$

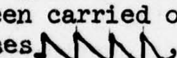
The factor in brackets is, in effect, an interpolation function: it approaches T/τ_c at small signal durations, and unity at large signal durations. There is precedent for forms similar to (5.8) in earlier work (e.g., Munson [45], Plomp and Bouman [43], Hughes [101], etc).

The Johnson-Robinson method [2], as generalized to incorporate equation (5.8), has been applied to the simulated blast waves dealt with in Fig. 30. Several choices of integration time τ_c were tried, and best fit to experiment was found with τ_c = 125 ms. This is labeled curve 3 in Fig. 32. For comparison, curve 2 is a calculation by the original Johnson-Robinson method, best fit being obtained by the choice τ_c = 175 ms.

In Fig. 33 the solid curve (experimental) and curve 3 (generalized Johnson-Robinson theory) are repeated from Fig. 32. Curve 2 on this figure is a calculation by the original Johnson-Robinson method with the choice τ_c = 125 ms. This curve 2 is clearly not a best fit to experiment, on an absolute basis. But the curve is a good fit on a relative basis: if it is downshifted by 1.3 dB it matches the solid curve well below T = 100 ms. In fact the downshifted τ_c = 125 ms curve 2 of Fig. 33 is virtually identical with the best-fit τ_c = 175 ms curve 2 of Fig. 32.

We have seen that the assumptions τ_c = 125 ms and τ_c = 175 ms both give comparably good relative fits to experiment (below their respective upper limits); this underlines the difficulty of inferring the correct physical critical integration time τ_c by such a matching procedure based on the linear integration assumption. The exponential integration assumption (5.8), with τ_c reinterpreted as an e^{-1} time constant, appears to yield a much more precise determination.

5.4.3 Conclusions

A set of equal-loudness comparison tests for idealized "quarry blast" waves has been carried out. The waveforms were simulated as a sequence of shock-decay pulses , the duration (~ number of shocks) being varied. The curve of overpressure ratio in phons vs duration shows a steep rise at small duration (< 100 ms, impulsive sound), but the curve bends over to approach an asymptotic upper limit at long duration (> 400 ms, continuous sound).

The Johnson-Robinson predictive theory for impulsive sound [2] gives good absolute agreement with the low end of the curve, provided the critical integration time of the ear τ_c is changed from 70 ms (their value) to 175 ms. Values of τ_c smaller than 175 ms give good relative agreement, but the curve is shifted upward. In this theory the band pressure levels are assumed to increase linearly with signal duration T via a factor (T/τ_c).

In our proposed generalization of the Johnson-Robinson theory the band pressure levels are assumed to follow an exponential saturation law $(1 - e^{-T/\tau_c})$. The generalized theory, with this redefined τ_c chosen as 125 ms, shows a very good fit to the empirical curve over the whole range. Thus it appears - at least for the "quarry blast" waveform - to integrate loudness prediction methods for transient and continuous sounds.

The question arises, can this generalized Johnson-Robinson theory be applied in hindsight to the earlier tests herein; will it correctly predict the loudness of sonic-boom wave-forms of the three kinds tested? The answer is yes, since the generalized and original Johnson-Robinson theories (with different choices for τ_c [see above]) agree fairly well at the smaller values of T/τ_c appropriate to sonic boom.

6. CONCLUDING REMARKS

The findings and conclusions drawn with respect to each of the four test series have been discussed in detail at the ends of the respective chapters. A summarized account is given verbally in the opening Abstract. An additional overview is given in schematic form in Table 6.1; the four waveforms tested are sketched at the head of four columns, and the respective key findings and conclusions are summarized underneath.

The results in each series were compared with theoretical predictions by the method of Johnson and Robinson. All but the long-duration quarry blast judgements were found to be in very good agreement in terms of relative loudness levels. With an ad hoc "exponential saturation" modification (including adjustment of the critical integration time of the ear) the predictive method was extended to encompass the long duration signals as well. Thus the applicability of the method has been demonstrated for other types of transient sounds than the N-wave; and the extension to the method appears to bridge the range between impulsive and continuous sounds of similar spectral content.

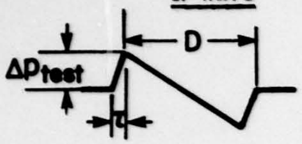
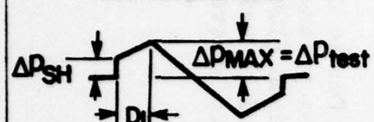
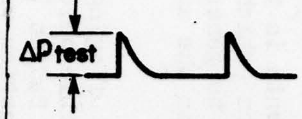
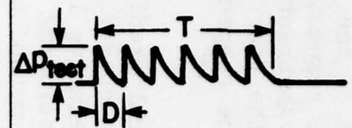
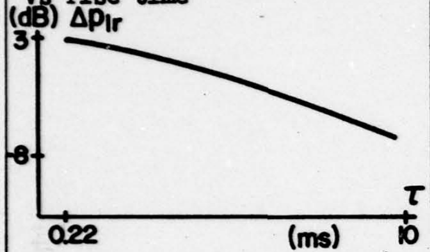
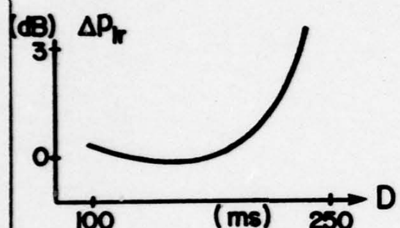
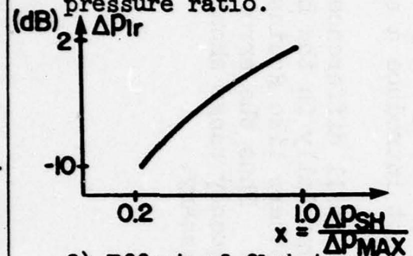
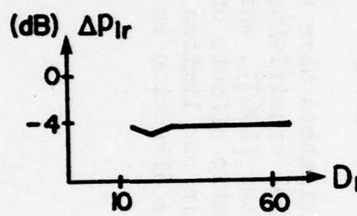
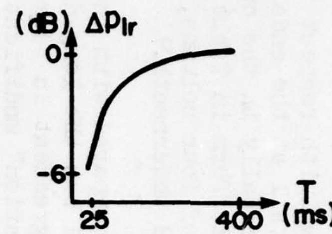
As stated earlier, the results for all loudness judgements in this project were acquired with an all male graduate student population (which imposes a limitation on age of about 21 to 30 years). In order to generalize the results we must know what is the influence of age and sex upon the auditory response to the transient sounds.

The reports in the literature indicate that the effect of aging upon the auditory system (i.e., presbycusis and sociocusis) is negligible below 2 kHz and is not of major significance below 5 kHz (Robinson and Dadson [104], Pollack [105], etc). The spectrum of test pulses in our investigation was regarded as significant only up to 5 kHz; consequently the generalization of our data over a wider age group should not introduce a serious error.

Some investigators have found small differences between female and male listeners in aural sensitivity, especially in the high frequency range (e.g., Rood [63], Bauer [106]), while others like Robinson and Dadson [104] have found no systematic effects of sex. Thus the error introduced by using only male observers in our limited frequency range should be small; however, more research in this direction is necessary.

Table 6.1 Subjective Comparison Tests - Summary of Results

[Definition: Overpressure level ratio $\Delta p_{1r} = -20 \log_{10}(\Delta p_{test}/\Delta p_{reference})_{\text{equal loudness}}$]

N-Wave	Flat-Top Boom	Filtered N-Wave	Idealized "Blast"
			
Equal-Loudness Contours 1) Overpressure level ratio vs rise time  2) Overpressure level ratio vs duration.  Conclusions 1) Very good agreement with previous work (except for duration 250 ms), especially with Johnson-Robinson theoretical calculations.	Equal-Loudness Contours 1) Effect of front shock overpressure/maximum overpressure ratio.  2) Effect of flat top duration.  Conclusions 1) Loudness dominated by front and rear shocks. 2) Empirical formula for calculation of N-wave overpressure which gives same subjective loudness. $\Delta p_N = \Delta p_{SH} + 0.11\ \Delta p_{MAX}$ 3) Very good agreement with calculated relative loudness (Johnson-Robinson theory).	Equal-Loudness Table 1) Very small decrease in subjective loudness of filtered N-wave (high pass filter f_c=25, 50 Hz) associated with reduction in low frequency content. 2) Very small increase in annoyance for filtered N-wave associated with reduction in low frequency content. Conclusions 1) Loudness of N-wave essentially insensitive to low frequency part of spectrum (up to about 50 Hz). 2) Very good agreement with calculated relative loudness (Johnson-Robinson theory).	Equal-Loudness Contour 1) Effect of duration.  Conclusions 1) Empirical curve shows temporal integration effect of human auditory system. Estimated critical time lies between 100 and 200 ms, as judged from knee of curve. 2) Very good agreement between empirical curve and calculated absolute loudness (basic Johnson-Robinson theory, with altered critical time τ_c=175 ms), over limited duration range ($25 \leq T \leq 100$ ms). 3) Proposed generalization of Johnson-Robinson theory gives very good agreement with empirical curve over full experimental range of durations ($25 \leq T \leq 400$ ms); thus it appears to integrate loudness prediction methods for transient and continuous sounds.

APPENDIX 1

EXPERIMENTAL RESULTS AND STATISTICAL ANALYSIS

N-WAVE COMPARISON TESTS

Table 1

Results of Sonic-Boom Comparison Experiment. Rise Time Variation

Reference Signature: Duration - 200 ms; Rise Time - 1 ms

Level Difference (dB) Required for Judged Equality

Rise time (ms)	Observer No.									
	1	2	3	4	5	6	7	8	9	10
0.22	3.5	4	4	4	1	4	2	3.3	3	2
0.5	3	5	4	3	1	2	2.7	2	2.5	2.9
1	0	0	-1	0	0.2	0	-1	0.5	0	1
3	-4	-4	-5	-4.6	-3.8	-5	-4.5	-6	-5.3	-5
10	-14	-6	-10	10	-6	-5	-5.5	-11	-6	-11

Rise time (ms)	Observer No.									
	11	12	13	14	15	16	17	18	19	20
0.22	4	4	2.5	2	3.2	2	2.8	4	1	2
0.5	1.3	2.7	2	2	2	3	0	3	1.2	2.8
1	-1	0	0	1	0.5	-1	1	-1	0	-1
3	-6	-5	-4	-4.7	-4.8	-5	-4	-5.2	-3	-5.2
10	-14	-8	-8	-9	-6.8	-8	-6	-4	-6	-0.4

Rise time (ms)	Overall Average All Observers (dB)
0.22	2.9
0.5	2.4
1	-0.09
3	-4.7
10	-7.8

Table 2

Standard Deviation for Data Points of the Graph Overpressure vs Rise Time

Rise Time (ms)	Standard Deviation (dB)
0.22	1.03
0.5	1.09
1	0.68
3	0.72
10	3.3

Table 3

One-Way Analysis of Variances for Rise Time Comparison

Source of Variation	Degrees of Freedom	Sum of Square	F
Between samples	4	SSB=1743.8	152.29
Error	95	SSE=270.92	
Total	99	SST=2014.72	

Table 4

Results of Sonic-Boom Comparison Experiment. Duration Variation

Reference Signature: Duration - 200 ms; Rise Time - 1 ms

Duration (ms)	Level Difference (+dB) Required for Judged Equality — Observer No. 1	2	3	4	5	6	7	8	9	10
100	1	1	0.7	2	1	-2	-2.4	1.3	1	0
150	-1	1	-0.8	-2	0	-2.5	-1	-2	1	-1
200	-1	-0.8	-2	0.7	-1.3	-0.7	0.5	-0.9	-0.9	1
250	3	3	2.7	2	3	1	5.2	6	4.5	4

Duration (ms)	Observer No. 11	12	13	14	15	16	17	18	19	20
100	-2	0.5	-1	0	0	-1	-1	0.5	0	-2.6
150	0.8	0	-1	-1.5	-0.5	-1.8	9	0.5	-1	-2
200	0	0	-1	-1	0	0	0.7	0	1	-0.8
250	3	0	2	3	3	5	2.8	4	2	2.9

Duration (ms)	Overall Average All Observers (dB)
100	0.15
150	-0.74
200	0.32
250	3.11

Table 5

Standard Deviation for Data Points

of the Graph Overpressure vs Duration

Duration (dB)	Standard Deviation (dB)
100	1.33
150	1.09
200	0.83
250	1.41

Table 6

One-Way Analysis of Variations

for Duration Comparison - Without Point D = 250 ms

Source of Variation	Degrees of Freedom	Sum of Square	F
Between samples	2	SSB=3.71	1.54
Error	57	SSE=68.71	
Total	59	SST=72.41	

APPENDIX 2

EXPERIMENTAL RESULTS AND STATISTICAL ANALYSIS

MINIMIZED SONIC BOOM COMPARISON TESTS

Table 1

Results of Minimized Low-Boom Comparison Experiment

Overpressure Ratio X Variation

Reference N-Wave: D = 150 ms; τ = 1 ms

Level Difference (+dB) Required for Judged Equality										
$x=\Delta p_{SH}/\Delta p_{MAX}$	Observer No.									
	1	2	3	4	5	6	7	8	9	10
0.2	-11	-11.25	-10.75	-11	-9.1	-11	-11.75	-11.25	-9.25	-9.5
0.4	-7	-5.25	-6.5	-5.25	-4.75	-6.75	-5	-5.5	-4.75	-6.5
0.6	-3.5	-1.25	-4	-5	0	-4	-2	-4.6	-3	-3
0.8	-1	1.35	-0.4	0	-1.25	-2	-0.75	-0.4	-0.8	-1
1.0	-0.5	3	1	0	1.5	0	2.5	0	3	2

$x = \Delta p_{SH}/\Delta p_{MAX}$	Overall Average All Observers (dB)
0.2	-10.48
0.4	-5.71
0.6	-3.03
0.8	-0.46
1.0	1.26

Table 2

Standard Deviation for Data Points

of the Graph Overpressure Level Ratio vs Parameter X

$x = \Delta p_{SH}/\Delta p_{MAX}$	Standard Deviation (dB)
0.2	0.85
0.4	0.86
0.6	1.56
0.8	0.99
1.0	1.34

Table 3

One-Way Analysis of Variance for Parameter X Comparison

Source of Variation	Degree of Freedom	Sum of Square	F
Between samples	4	SSB=855.06	160.56
Error	45	SSE= 59.91	
Total	49	SST=914.97	

Table 4

Results of Minimized Low-Boom Comparison Experiment

Flat Top Duration D_1 Variation

Reference N-Wave: D = 150 ms; τ = 1 ms

Level Difference (+dB) Required for Judged Equality							
D_1 (ms)	Observer No.						
	1	2	3	4	5	6	10
10	-4.75	-4.75	-5	-4	-4.75	-6	-4
20	-8	-4	-4.5	-6	-4	-5	-5.25
30	-5	-4.75	-5	-2	-4.25	-5	-4
40	-4.5	-5.25	-4	-5.25	-3.25	-5	-4
60	-3	-5	-6	-5.25	-3.1	-5.5	-4

D_1 (ms)	Overall Average All Observers (dB)
10	-4.75
20	-5.25
30	-4.28
40	-4.46
60	-4.55

Table 5

Standard Deviation for Data Points

of the Graph Overpressure Level Ratio vs Flat Top Duration

D_1 (ms)	Standard Deviation (dB)
10	0.68
20	1.41
30	1.08
40	0.76
60	1.19

Table 6

One-Way Analysis of Variance for Flat Top Duration Comparison

Source of Variation	Degree of Freedom	Sum of Square	F
Between samples	4	SSB= 3.83	0.85
Error	30	SSE=33.61	
Total	34	SST=37.44	

APPENDIX 3

EXPERIMENTAL RESULTS AND STATISTICAL ANALYSIS

FILTERED N-WAVE COMPARISON TESTS

Table 1

Results of Filtered N-Wave Comparison Experiment

Observer No.	Level Difference (+dB) Required for Judged Equality			
	Loudness		Annoyance	
	f_c = 25 Hz	f_c = 50 Hz	f_c = 25 Hz	f_c = 50 Hz
1	-0.5	-1.3	0	-0.5
2	0	3	2.2	1
3	0	0		
4	0	0	-1	1
5	-1	-0.75		
6	-1.25	-2.25	2	0
7	1	-0.75	0	1
8	0.4	-1	0	2
9	1	-1		
10	0	0		
11	1	0	0	2
12	0	-1	1	0
13	0.75	-0.75		
14	-0.75	0.8		
15	-2	-1.25	0	0
16			-1	1
Overall Average	-0.09	-0.42	0.52	0.55

Table 2

Standard Deviation for Data

from the Filtered N-Wave Comparison Experiment

f_c (Hz)	Standard Deviation (dB)
Loudness	
25	0.89
50	1.36
Annoyance	
25	1.01
50	1.01

Table 3

Test of Significance "t" for Results of Filtered N-Wave Comparison Experiment

Difference Between Mean Values for 25 and 50 Hz Cut-Off Frequency Case

	Loudness Comparison	Annoyance Comparison
Degrees of Freedom	df = 14	df = 9
Standard deviation of difference	$s_d^2 = 1.809$	$s_d^2 = 2.63$
t value	t = -0.94	t = 0.06

Table 4

Test of Significance "t" for Results of Filtered N-Wave Comparison Experiment

Difference Between Mean Values for Loudness and Annoyance Comparisons

	Cut-Off Frequency	
	$f_c = 25$ Hz	$f_c = 50$ Hz
Number of variates	$n_1 = 15$ $n_2 = 10$	$n_1 = 15$ $n_2 = 10$
Standard deviation $s^2 = \frac{x}{n - 1}$	$s_1^2 = 0.78$ $s_2^2 = 1.02$	$s_1^2 = 1.43$ $s_2^2 = 1.03$
t value: $t = \frac{\bar{X}_1 - \bar{X}_2}{\sqrt{\frac{s_1^2}{n_1} + \frac{s_2^2}{n_2}}}$	$t = 1.56$	$t = 2.17$
t value required for significance $t' = \frac{t_1 \frac{s_1^2}{n_1} + t_2 \frac{s_2^2}{n_2}}{\frac{s_1^2}{n_1} + \frac{s_2^2}{n_2}}$	$t' = 1.81$ at 10%	$t' = 2.2$ at 5% $t' = 1.8$ at 10%

where x - variates

$\bar{X}_{1,2}$ - mean values

$t_{1,2}$ - t values taken from "t" table for $(n_1 - 1)$ and $(n_2 - 1)$ degrees of freedom

APPENDIX 4

EXPERIMENTAL RESULTS AND STATISTICAL ANALYSIS

IDEALIZED "BLAST" SIGNATURE COMPARISON TESTS

Table 1

Results of Blast Signature Comparison Experiment

Observer No.	Level Difference (+dB) Required for Judged Equality			
	Ref. D = 50 ms Test D = 25 ms	Ref. D = 100 ms Test D = 50 ms	Ref. D = 200 ms Test D = 100 ms	Ref. D = 400 ms Test D = 200 ms
1	2	3	3	1
2	1.3	4	3	1
3	1.35	2	1	0.75
4	1	1	0	0.33
5	2	2.65	0	-1
6	2	2	1.25	0.75
7	4	3	-1	0.65
8	1	4	2.75	0
9	1	3	3	2
10	1	1	0	0.65
11	1.2	2.35	1.35	2
12	3	2	2	1.25
13	1	3.35	2	1
14	0	2.75	-1	0
15	2	2	2.8	0.33
Overall Average	1.59	2.54	1.34	0.71

Table 2

Standard Deviation for Data

from Blast Signature Comparison Experiment

D_R (ms) - D_T (ms)	Standard Deviation (dB)
50 - 25	0.96
100 - 50	0.91
200 - 100	1.46
400 - 200	0.76

Table 3

One-Way Analysis of Variance for Blast Signature Duration Comparison

(Re 400 ms case taken as 0 dB)

Source of Variation	Degree of Freedom	Sum of Square	F
Between samples	3	SSB=267.77	22.14
Error	56	SSE=225.75	
Total	59	SST=493.52	

APPENDIX 5

INSTRUCTION AND CONSENT FORM FOR EXPERIMENTATION

UTIAS SONIC BOOM PROJECT

"CONSENT"

I, ____ consent to take part in a research project being carried out at the University of Toronto, Institute for Aerospace Studies, under the direction of Professor H. S. Ribner. I understand that my participation will include the following:

(a) I will be exposed to pairs of transient pressure signals (simulated sonic booms and other similar pulses) while seated in the UTIAS sonic boom simulation booth.

(b) I will compare the relative loudness (or annoyance) of the paired transient pressure signals.

(c) Each session will be no longer than 15 minutes and there will be no more than two sessions per day, totalling up to six sessions per test.

(d) A qualified otolaryngologist will examine my hearing before the experiment and in addition an audiometric test will be performed before and after each session.

I understand that I may withdraw from the project at any time.

Dated at Toronto this____ day of____________197__.

______________________________ ______________________________

Witness Signature

INSTRUCTIONS

The purpose of the test is primarily to determine the loudness of different transient sounds as a function of some particular features of those pulses.

You will hear a series of impulsive sounds while seated inside the booth. The sounds will occur in "pairs" and will be separated by few seconds. Your task will be to judge which sound in each pair is louder. After each pair of signals there will be an interval of silence so you will have enough time to communicate your judgement through intercom.

Please base your decisions on your subjective feeling of loudness, disregarding other differences between pulses which may occur. In order to help you in your task, five standard scores will be used:

first (in pair) louder

first (in pair) may be louder

both equal

second (in pair) louder

second (in pair) may be louder

Please concentrate on the judgement. There is no "good" or "bad" answer; we are just interested in your subjective feeling. Take this experiment seriously, otherwise it will be a waste of time for you as well as for us.

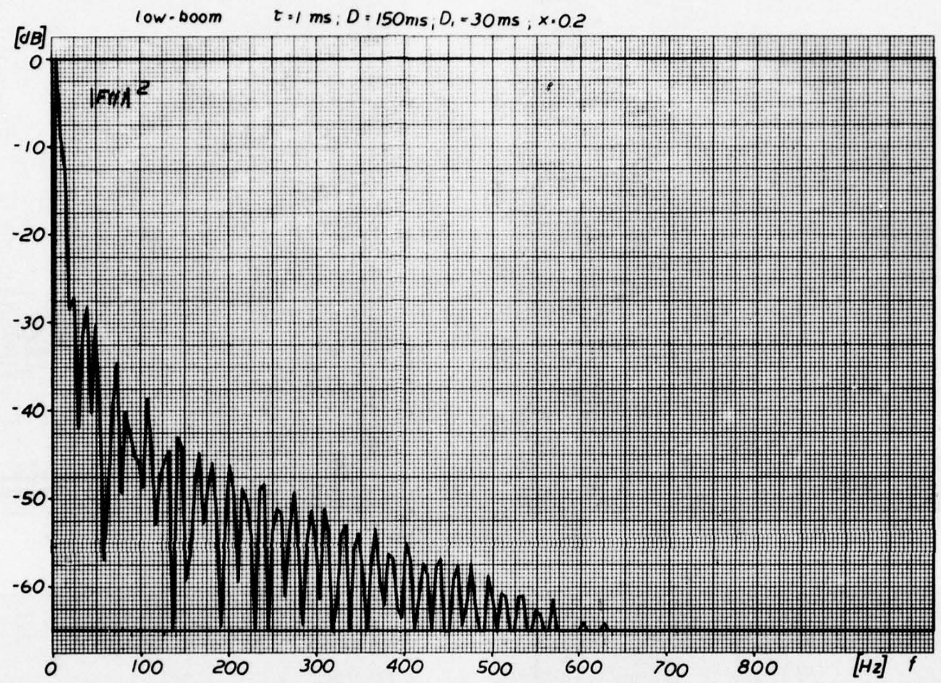
APPENDIX 6

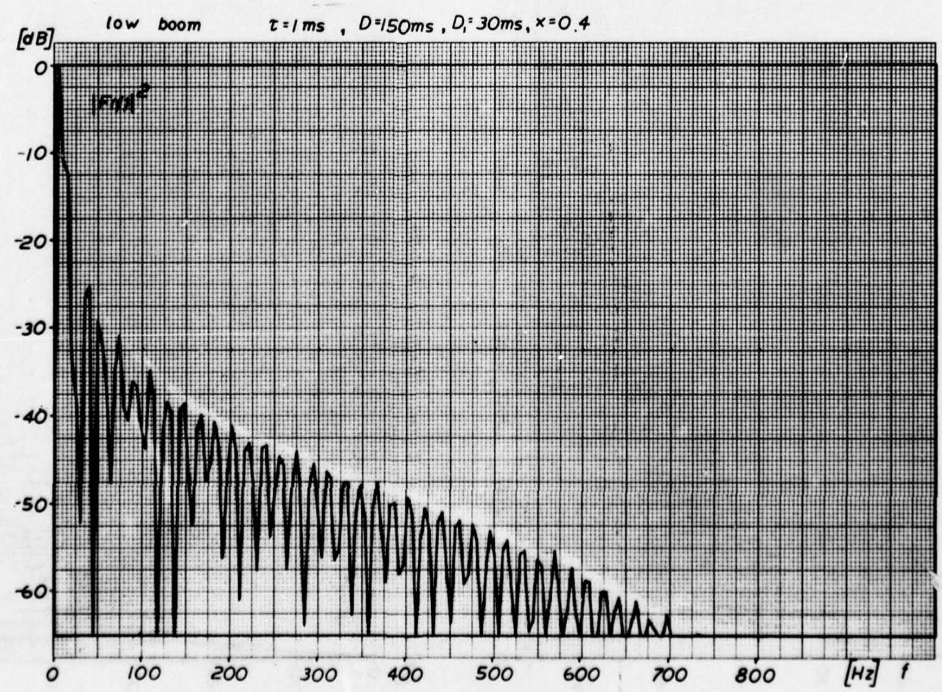
EXAMPLES OF COMPUTER-CALCULATED SPECTRAL ENERGY DENSITIES

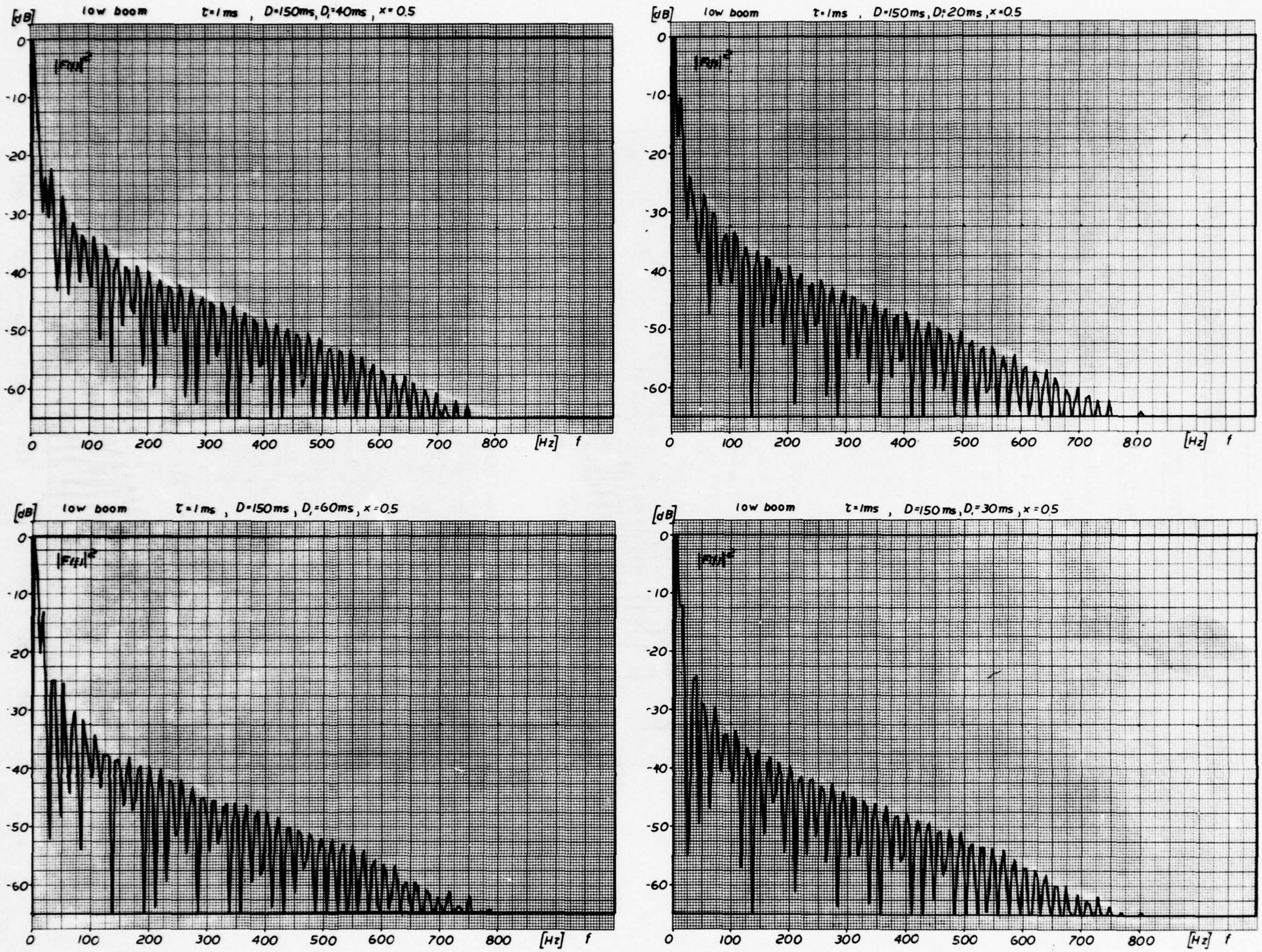
OF TEST SIGNALS

Part 1(a)

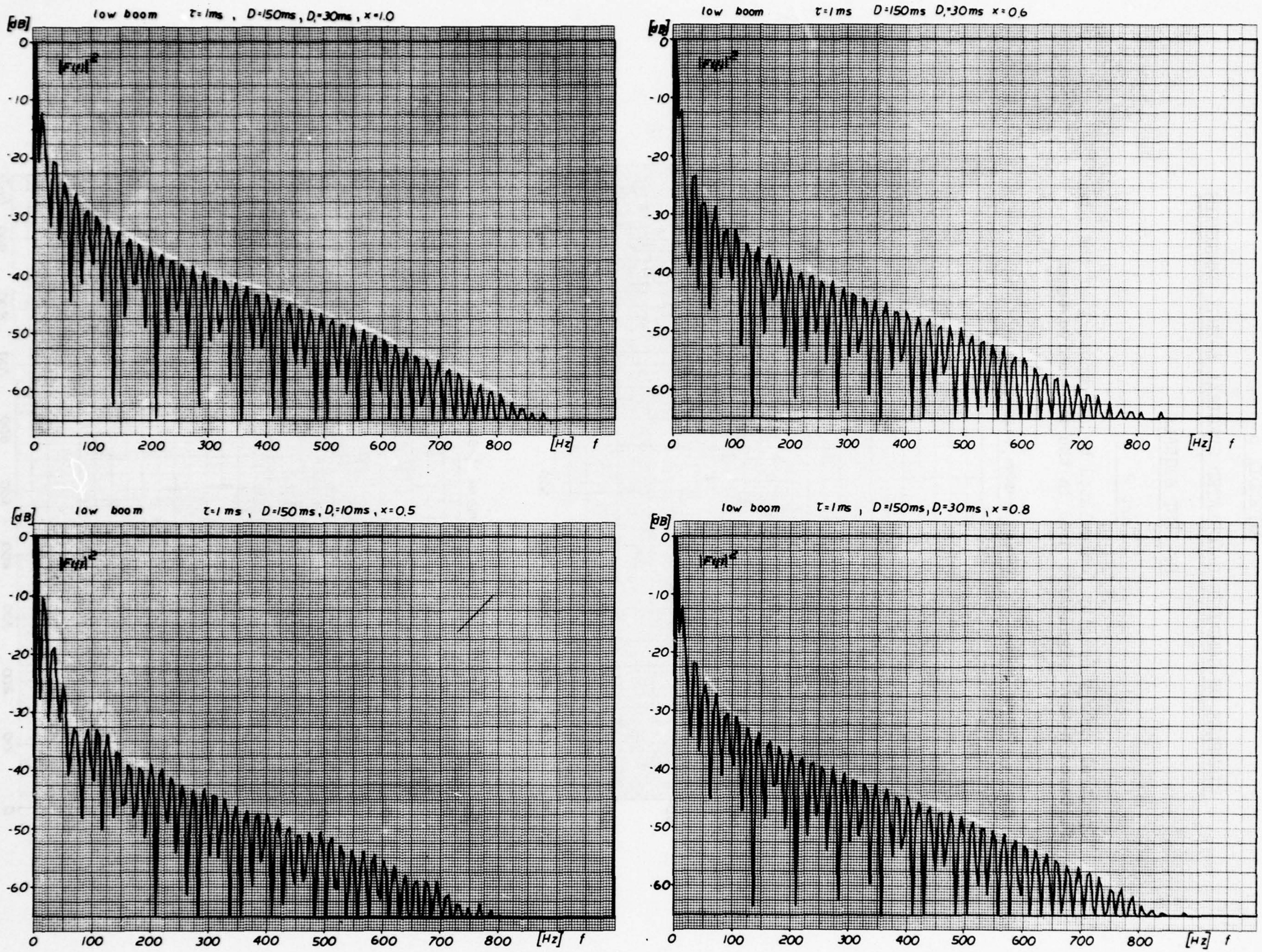
Spectral energy density (calculated by a computer) of the ideal "low-boom" signatures used in the subjective test.







APPENDIX 6 - Part 1(b)



APPENDIX 6 - Part 1(c)

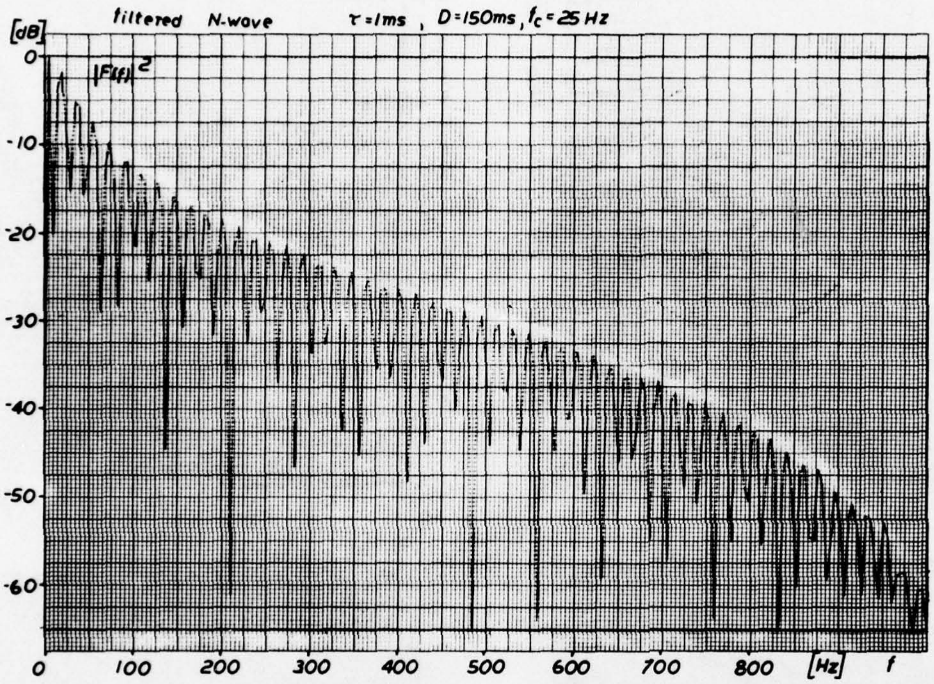
APPENDIX 6

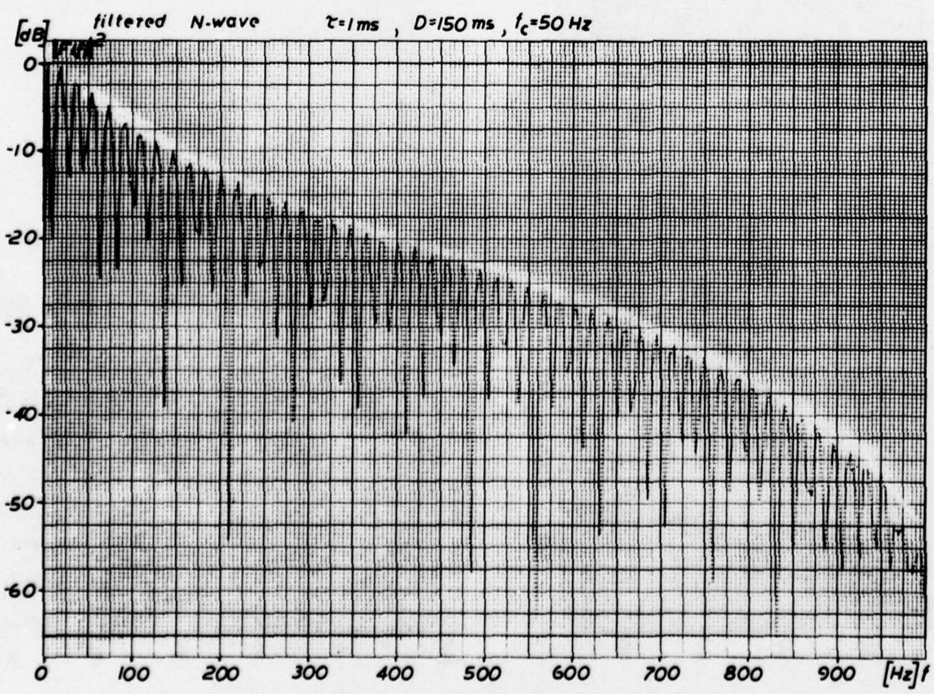
EXAMPLES OF COMPUTER-CALCULATED SPECTRAL ENERGY DENSITIES

OF TEST SIGNALS

Part 2

Spectral energy density (calculated by a computer) of the ideal filtered N-wave signatures used in the subjective test.



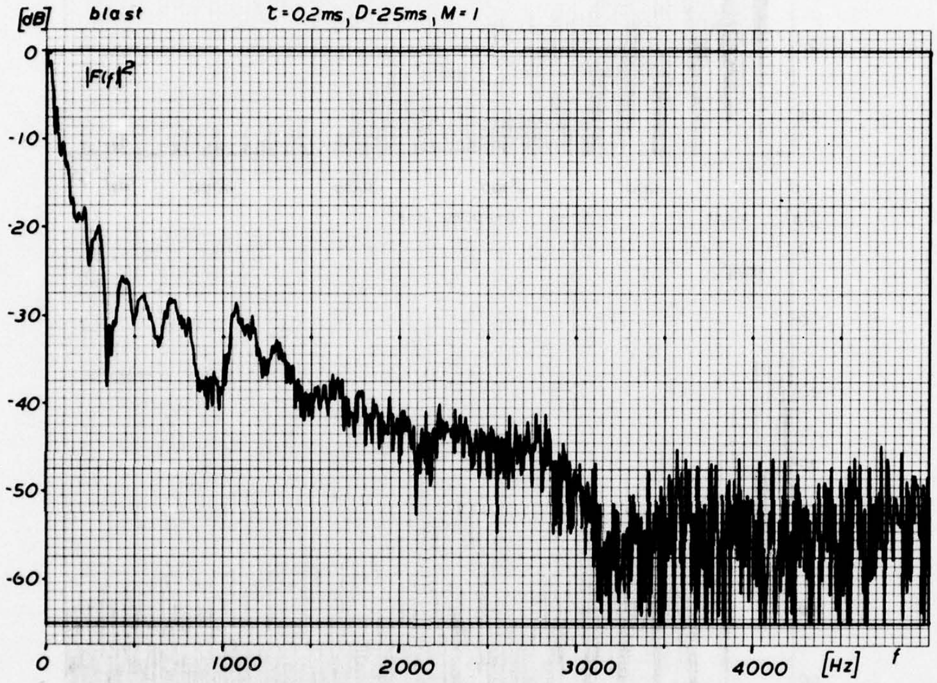


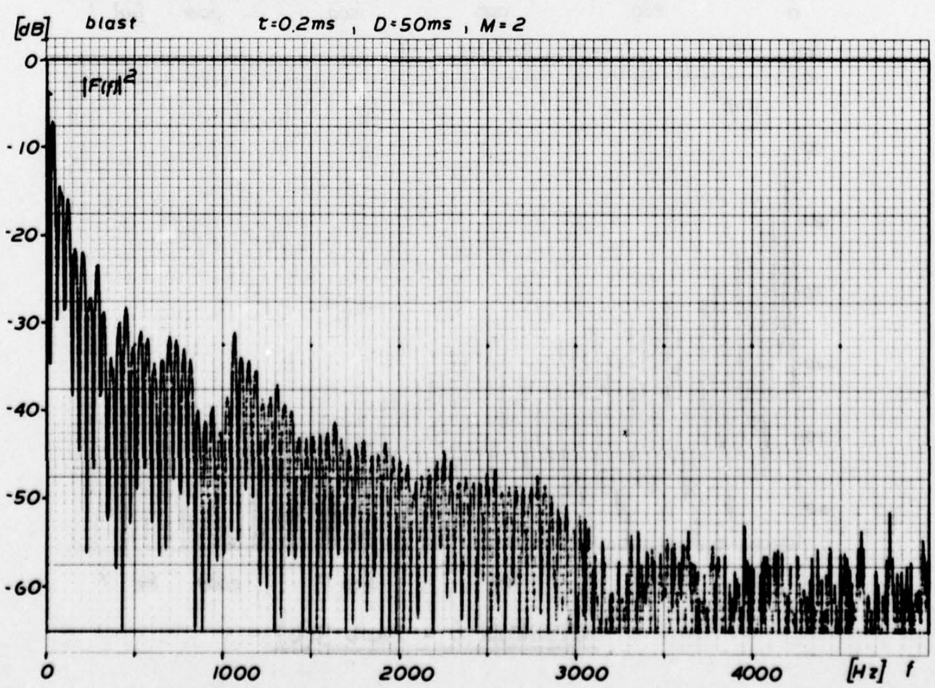
APPENDIX 6

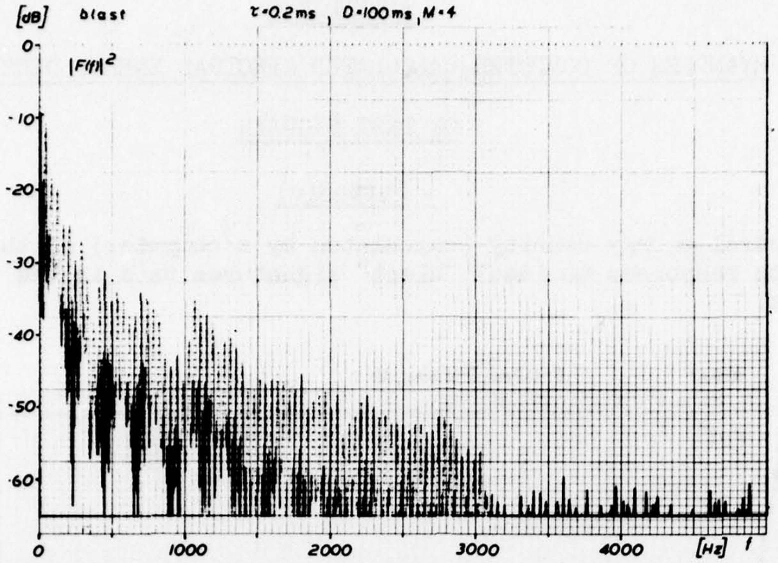
EXAMPLES OF COMPUTER-CALCULATED SPECTRAL ENERGY DENSITIES OF TEST SIGNALS

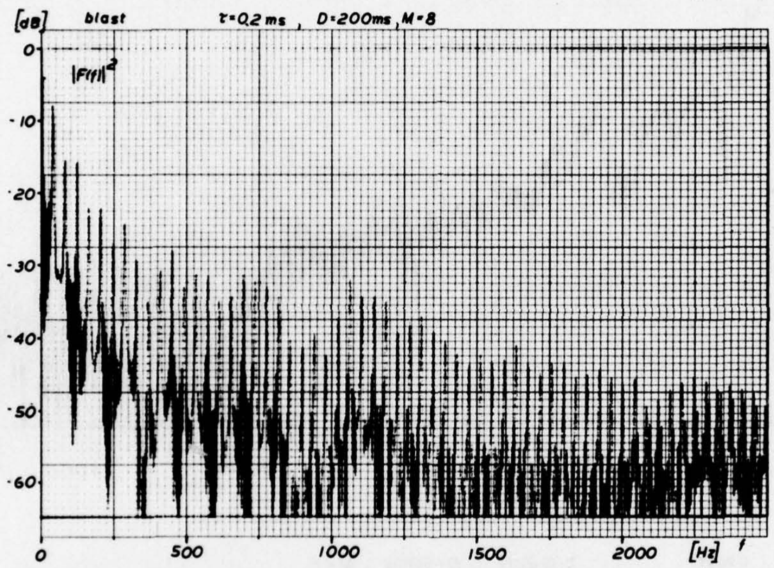
Part 3(a)

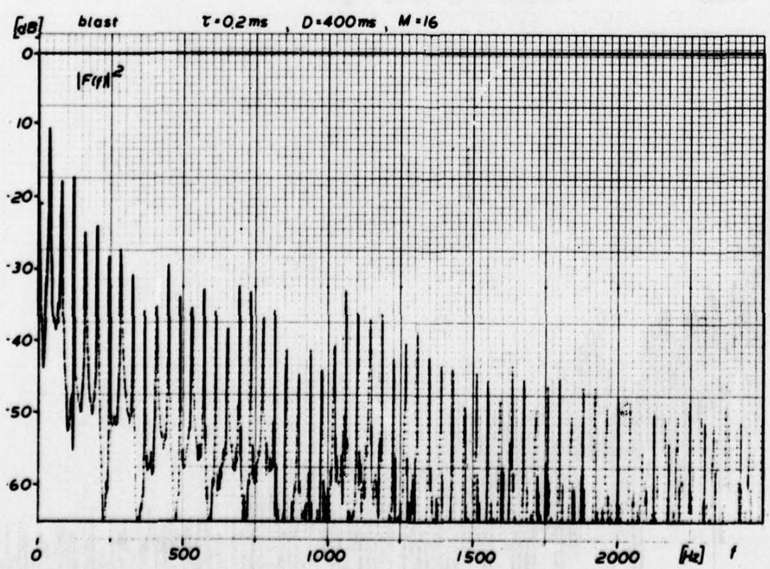
Spectral energy density (calculated by a computer) of the recorded booth responses to ideal "blast" signatures used in the subjective test.











APPENDIX 6 - Part 3(b)

APPENDIX 7

COMPUTER PROGRAMS

Part 1

Listing of the FFT Fortran program (IBM 360) calculating decibel values of one-third octave band pressure levels using the Johnson-Robinson method [2]. Program reads input data (signatures stored in digital form by the HP 2100 computer with A/D converter) from a magnetic tape, calculates Fourier transform using a standard FFT subprogram and computes energy in one-third octave bands. These energies are converted into pressure levels (using the Johnson-Robinson scheme) and printed as output data.

```
C     FFT
      DIMENSION A(8192),AC(4096),IWK(14)
      DIMENSION O(34),E(33)
      INTEGER*2 I1(9000)
      COMMON /A1/I1
      COMPLEX GMN,AC
      EQUIVALENCE  (AC(1),A(1))
      DATA O/4.4,5.6,7.0,8.9,11.2,14.1,17.8,22.4,28.2,35.5,44.7,56.2,
     *70.8,89.1,112.0,141.0,178.0,224.0,282.0,355.0,447.0,562.0,708.0,
     *891.0,1122.0,1413.0,1778.0,2239.0,2818.0,3548.0,4467.0,5623.0,
     *7079.0,8913.0/
      KT=0
      READ(5,1) NRT
1     FORMAT(I4)
2     FORMAT (4I4)
3     FORMAT(3F7.2)
100   IF(KT-NRT) 1000,1001,1000
1000  READ(5,2)NRI,NP,NPI,LS
      READ(5,3) TT,T,TX
      READ(5,6)P
6     FORMAT(F9.4,
      NX=TT/T
      AX=T/0.05
      NY=IFIX(TX)

      NN=0
      CALL READI1(20,9000,NRI,ICODE)
      IF(ICODE.EQ.-1) GO TO 995
      AL=10.0
      DO 121 I=1,NX
      AL=AL+AX
      MY=AL
121   A(I)=I1(MY)
      CALL FFTR(A,GMN,NY,IWK)
      NX=NY/2
      F=1000.0/(TX*T)
      DO 210 I=1,NX
210   A(I)=(CABS(AC(I))*0.05E-3*P/2047.0)**2
      A(NX+1)=(CABS(GMN)*0.05E-3*P/2047.0)**2
      DO 800 K=1,33
      NB=O(K)/F
      NBL=NB+1
      NE=O(K+1)/F
      NEL=NE+1
      YB=A(NBL)+(A(NBL+1)-A(NBL))*(O(K)/F-FLOAT(NB))
      YE=A(NEL)+(A(NEL+1)-A(NEL))*(O(K+1)/F-FLOAT(NE))
      IF(NE-NB)51,51,50
51    E(K)=(O(K+1)-O(K))*(YB+YE)/2.0
      GO TO 800
50    M=NB+1
      E(K)=(FLOAT(M)*F-O(K))*(YB+A(NBL+1))/2.0
54    M=M+1
      FX=FLOAT(M)*F
      IF(FX-O(K+1))52,53,64
52    E(K)=E(K)+F*(A(M+1)+A(M))/2.0
      GO TO 54
53    E(K)=E(K)+F*(A(M)+A(M+1))/2.0
      GO TO 800
64    E(K)=E(K)+(O(K+1)-FLOAT(NE)*F)*(YE+A(NEL))/2.0
800   CONTINUE

      DO 70 K=1,33
      E(K)=10.0*ALOG10((2.0*E(K))/(4.0E-8*0.07))-3
      KK=K+6
      WRITE(6,7)KK,E(K)
7     FORMAT(1X,8(I2,'-',F8.2))
70    CONTINUE
705   KT=KT+1
      GO TO 100
1001  CONTINUE
995   STOP
       END

      SUBROUTINE READI1(IUNIT,LEN,IREC,ICODE)
      INTEGER*2 I1(9000)
      COMMON/A1/I1
      ICODE=0
      NREC=LEN/250
      IF(NREC*250 .NE. LEN) GO TO 999
      REWIND IUNIT
      NSKP=NREC*(IREC-1)
      IF (NSKP .EQ. 0) GO TO 1
       DO 5 I=1,NSKP
      READ(IUNIT,103,END=998) DUMMY
103   FORMAT(A2)
5     CONTINUE
1     CONTINUE
      II=1
      IL=250
      DO 10 KI=1,NREC
      READ(IUNIT,101,END=998)(I1(I),I=II,IL)
101   FORMAT(250A2)
      II=IL+1
      IL=IL+250
10    CONTINUE
      RETURN
C     E O F ERROR
998   CONTINUE
      WRITE(6,102)IUNIT
102   FORMAT('  END OF FILE ON UNIT',I5)
      ICODE=-1
      RETURN
C     INCORRECT LENGHT ERROR
999   CONTINUE
      WRITE(6,100)LEN
100   FORMAT('  INVALID LENGHT=',I10,' NOT A MULTIPLE OF 250')
      ICODE=-1
      RETURN
      END
```

```
      FTN,B
            PROGRAM PRDST
      C     PREDEST GENERATES THE PREDESTORED IMPULSES FROM IMPULS
      C     AND DESTORTING CHARACTERISTIC.RESULT IS STORED ON
      C     MAGNETIC TAPE IN 2048 WORD ARRAY.
            DIMENSION ISCAL(2),ISTAT(2),IDATA(2048,2),I1(5000),
           *DATA(2048,3)
            EQUIVALENCE(I1(905),DATA(1,1)),(IDATA(1,1),DATA(1024,2))
            LEN=2048
            CALL CFG71(LEN,IDATA,ISCAL,ISTAT)
            MF=1
            MR=0
      80    NR1=0
            KT=0
            READ(1,6)NRT
      6     FORMAT(I4)
            READ(1,*)DUR
      100   IF(KT-NRT)1000,1001,1001
      1000  L=1
            READ(1,1)NFB,NRB,NFN,NRN,NFG,NRG
      1     FORMAT(6I4)
            LO=0
            M=1
            N=3
            KL=2
            LL=2
      101   IF(L-2)102,103,126
      102   NF=NFB
            NR=NRB
            GO TO 50
      103   NF=NFG
            NR=NRG
            LL=1
            GO TO 50
      123   NF=NFN
            NR=NRN
            KL=1
            LL=1
            M=3
            N=2
      50    IF(NF-MF)104,110,111
      104   J=MF+1-NF
            DO 105 I=1,J
      105   CALL BAAAL
            IF(MF-1)107,107,106
      106   CALL BAAAI
      107   J=NR-1
            GO TO 113
      110   J=NR-MR-1
            GO TO 113
      111   J=NF-MF
            DO 112 I=1,J
      112   CALL BAAAD
            J=NR-1
      113   IF(J) 114,120,116
      114   J=-J
            DO 115 I=1,J
      115   CALL BAAAK
            GO TO 120
      116   DO 117 I=1,J
      117   CALL BAAAI
      120   MF=NF
            MR=NR
            IF(LO)220,220,145
      220   CONTINUE
            CALL BAAAE(5000,I1)
            A=0.0
            K=0
            AX=0.0
            AN=DUR/0.05
            AX=AN/2048.0
            DO 121 I=1,2048
            A=A+AX
            K=IFIX(A)+10
            IDATA(I,KL)=(FLOAT(I1(K))/2047.0)*32767.0
      121   CONTINUE
            ISCAL(KL)=0
            ISTAT(KL)=0
            CALL XEQ71(17,KL)
      122   CALL XEQ71(00,KL)
            DO 1220 I=1,LEN
      1220  DATA(I,LL)=FLOAT(IDATA(I,KL))*(2.0**ISCAL(KL))
            WRITE(2,2) L
      2     FORMAT(I4,"FFT")
            L=L+1
      124   GO TO 101
      126   CONTINUE
      131   DO 132 I=1,2
            IF(DATA(I,M))132,150,132
      150   DATA(I,M)=1.0
      132   DATA(I,N)=DATA(I,KL)/DATA(I,M)
            DO 133 I=3,LEN,2
            NO=I+1
            AM=DATA(I,M)**2+DATA(NO,M)**2
            IF(AM)205,206,205
      206   DATA(I,M)=1.0
            DATA(NO,M)=1.0
      205   DATA(I,N)=(DATA(I,KL)*DATA(I,M)+DATA(NO,KL)*DATA(NO,M))/
           *(DATA(I,M)**2+DATA(NO,M)**2)
            DATA(NO,N)=(DATA(I,M)*DATA(NO,KL)-DATA(I,KL)*DATA(NO,M))/
           *(DATA(I,M)**2+DATA(NO,M)**2)
      133   CONTINUE
            IF(L-3)123,123,135
      135   AMAX=ABS(DATA(1,2))
            DO 137 I=1,LEN
            D=ABS(DATA(I,2))
            IF(AMAX-D)136,137,137
      136   AMAX=D
      137   CONTINUE
            DO 138 I=1,LEN
      138   IDATA(I,2)=(DATA(I,2)/AMAX)*32767.0
            ISCAL(2)=0
            ISTAT(2)=0
            CALL XEQ71(17,2)
            CALL XEQ71(01,2)
            WRITE(2,3)
      3     FORMAT("INV.FFT")
            A=IDATA(1,2)
            AMAX=ABS(A)
            DO 139 I=1,LEN
            A=IDATA(I,2)
            D=ABS(A)
            IF(AMAX-D)140,139,139
      140   AMAX=D
      139   CONTINUE
            DO 143 I=1,2048
      141   IDATA(I,1)=(FLOAT(IDATA(I,2))/AMAX)*2047.0
      143   CONTINUE
            READ(1,5)NFO,NRO
      5     FORMAT(2I4)
            NF=NFO
            NR=NRO
            LO=1
            GO TO 50
      145   CONTINUE
            AX=2048.0/AN
            A=AX
            DO 304 I=1,10
      304   I1(I)=0
            DO 300 I=11,5000
            K=A
            IF(K-1)301,302,302
      301   I1(I)=IDATA(1,1)
            GO TO 303
      302   IF(K-2048)306,306,305
      305   I1(I)=0
            GO TO 303
      306   I1(I)=IDATA(K,1)
      303   A=A+AX
      300   CONTINUE
            CALL BAAAH(5000,I1)
            KT=KT+1
            GO TO 100
      1001  CONTINUE
            CALL BAAAJ
            GO TO 80
            END
            END$
```

APPENDIX 7

COMPUTER PROGRAMS

Part 2(a)

Listing of the PRSDT Fortran program (HP 2100) with assembler subroutine controlling the magnetic tape movement. The program calculates a transfer function from two input signatures (see Chapter 3.3), alters a complex spectrum of a third signature by dividing it by the transfer function and transforms the result back into the time domain. All input and output signatures are stored on a magnetic tape in digital form. Fourier transforms are performed using the Fast Transform Arithmetic Unit, Model 5471A, being part of an additional equipment of the computer.

```
ASMB,R,B,L
*       TAPE CONTROLLER
        NAM  BAAAD
        ENT  BAAAD,BAAAE,BAAAF,BAAAH
        ENT  BAAAI,BAAAJ,BAAAK,BAAAL
        EXT  .ENTR,.IOC.
*       FORWARD SPACE 1 FILE ( FSF )
BAAAD   NOP
        ISZ  BAAAD
        JSB  .IOC.    FSF
        OCT  32007
        JMP  *-2
        JSB  .IOC.    WAIT FOR COMPLETION
        OCT  30007
        SSA
        JMP  *-3
        JMP  BAAAD,I
*       BINARY TAPE READ ( RRF )
E       BSS  2
BAAAE   NOP
        JSB  .ENTR
        DEF  E
        LDA  E,I
        STA  E1
        JSB  .IOC.    RRF
        OCT  10107
        JMP  *-2
        DEF  E+1,I
E1      NOP
        JSB  .IOC.    WAIT FOR COMPLETION
        OCT  30007
        SSA
        JMP  *-3
        AND   MASK
        SZA
        HLT   66B
        JMP  BAAAE,I
*       REWIND TAPE ( REW )
F       OCT  100
BAAAF   NOP
        ISZ  BAAAF
        JSB  .IOC.    REW
        OCT  30407
        JMP  *-2
        JSB  .IOC.    WAIT TIL TAPE STOPS THEN
        OCT  30007    SENSE BOT
        SSA
        JMP  *-3
        AND  F
        SZA,RSS
        JMP  BAAAF+2
        JMP  BAAAF,I
*       BINARY TAPE WRITE ( WCC )
H       BSS  2
BAAAH   NOP
        JSB  .ENTR
        DEF  H
        LDA  H,I
        STA  H1
        JSB  .IOC.    WCC
        OCT  20107
        JMP  *-2
        DEF  H+1,I
H1      NOP
        JSB  .IOC.    WAIT FOR COMPLETION
        OCT  30007
        SSA
        JMP  *-3
        AND   MASK
        SZA
        HLT   55B
        JMP  BAAAH,I
MASK    OCT   377
*       FORWARD SPACE 1 RECORD (FSR )
BAAAI   NOP
        ISZ  BAAAI
        JSB  .IOC.    FSR
        OCT  30307
        JMP  *-2
        JSB  .IOC.    WAIT FOR COMPLETION
        OCT  30007
        SSA
        JMP  *-3
        JMP  BAAAI,I
*       WRITE END OF FILE (EOF)
BAAAJ   NOP
        ISZ  BAAAJ
        JSB  .IOC.    EOF
        OCT  30107
        JMP  *-2
        JSB  .IOC.    WAIT FOR COMPLETION
        OCT  30007
        SSA
        JMP  *-3
        JMP  BAAAJ,I
*       BACK SPACE ONE RECORD ( BSR )
BAAAK   NOP
        ISZ  BAAAK
        JSB  .IOC.    BSR
        OCT  30207
        JMP  *-2
        JSB  .IOC.   WAIT FOR COMPLETION
        OCT  30007
        SSA
        JMP  *-3
        JMP  BAAAK,I
*       BACK SPACE 1 FILE (BSF)
BAAAL   NOP
        ISZ  BAAAL
        JSB  .IOC.    BSF
        OCT  32107
        JMP  *-2
        JSB  .IOC.   WAIT FOR COMPLETION
        OCT  30007
        SSA
        JMP  *-3
        JMP  BAAAL,I
        END
```

APPENDIX 8

JOHNSON AND ROBINSON METHOD FOR CALCULATING LOUDNESS OF IMPULSIVE SOUNDS:

STEP-BY-STEP COMPUTER PROCEDURE

1. The measured impulsive sounds recorded on an FM tape recorder are converted to digital form by an A/D converter and stored in a magnetic tape storage (HP 2100A). The frequency of sampling is $f_s = 20$ kHz. The maximum amplitude is scaled to ± 2047 discrete points.

2. The Fast Fourier subroutine computes the complex discrete values of Fourier components for k = 0, 1, ..., N-1 defined as:

$$X_k = \sum_{n=0}^{N-1} x_n \exp\left[-j\frac{2\pi kn}{N}\right] \qquad (1)$$

where

$$x_n = x(n \cdot \Delta t)$$

$$X_k = \frac{X(f_k,T)}{\Delta t}$$

$$f_k = k \cdot \Delta f = \frac{k}{T} = \frac{k}{N \cdot \Delta t}$$

T = total duration of sample

Δt = sampling time

N = number of discrete points in sample

$\frac{1}{T} = \Delta f$, frequency separation in frequency domain

3. The energy in 1/3 octave bands is calculated by summing up all discrete values of the energy spectral density $|X_k \cdot \Delta t|^2$ multiplied by the frequency separation $\Delta\omega = 2\pi \cdot \Delta f$:

$$E_b = \frac{1}{\pi}\sum_{k=A}^{B} |X_k \cdot \Delta t|^2 \; \Delta\omega \qquad (2)$$

where A and B are discrete limits of a 1/3 octave band.

4. The effective 1/3 octave band pressure levels are computed from band energies:

$$L_p = 10 \log_{10} \frac{E_b}{p_{ref}^2 \cdot t_c} \tag{3}$$

where

$$p_{ref} = 0.0002 \ \mu bar$$

$$t_c = \begin{cases} \tau_c & \text{- original Johnson-Robinson procedure} \\ \dfrac{T_e}{1 - e^{-T_e/\tau_c}} & \text{- generalized Johnson-Robinson procedure} \end{cases}$$

T_e - effective duration of the signal

τ_c - critical time

5. In the case of sonic booms with duration $\geq$ 100 ms the front and rear shocks are normally heard separately ("bang-bang"). For this circumstance Johnson and Robinson reduce the pressure levels in 1/3 octave bands by 3 dB:

$$L_p' = L_p - 3 \text{ dB} \tag{4}$$

This correction is restricted to sonic booms, or double impulses of similar duration.

6. Levels in bands below 50 Hz are reduced to levels at this frequency (i.e., 50 Hz) which produce the same effect (same loudness in phons); equal loudness curves are used, e.g., ISO Recommendation R226 [27]. These weighted levels are then combined with that already existing in the 50 Hz band, using normal rules of decibel addition.

7. The resulting modified effective pressure levels in 1/3 octave bands are used for evaluating the subjective loudness. The standard loudness calculation procedure by Stevens (ISO Recommdnation R532 [30]) was used in the present work. The loudness in sons (S_t) is converted to loudness level in phons (P) by the equation

$$P = 40 + 10 \log_2 S_t \tag{5}$$

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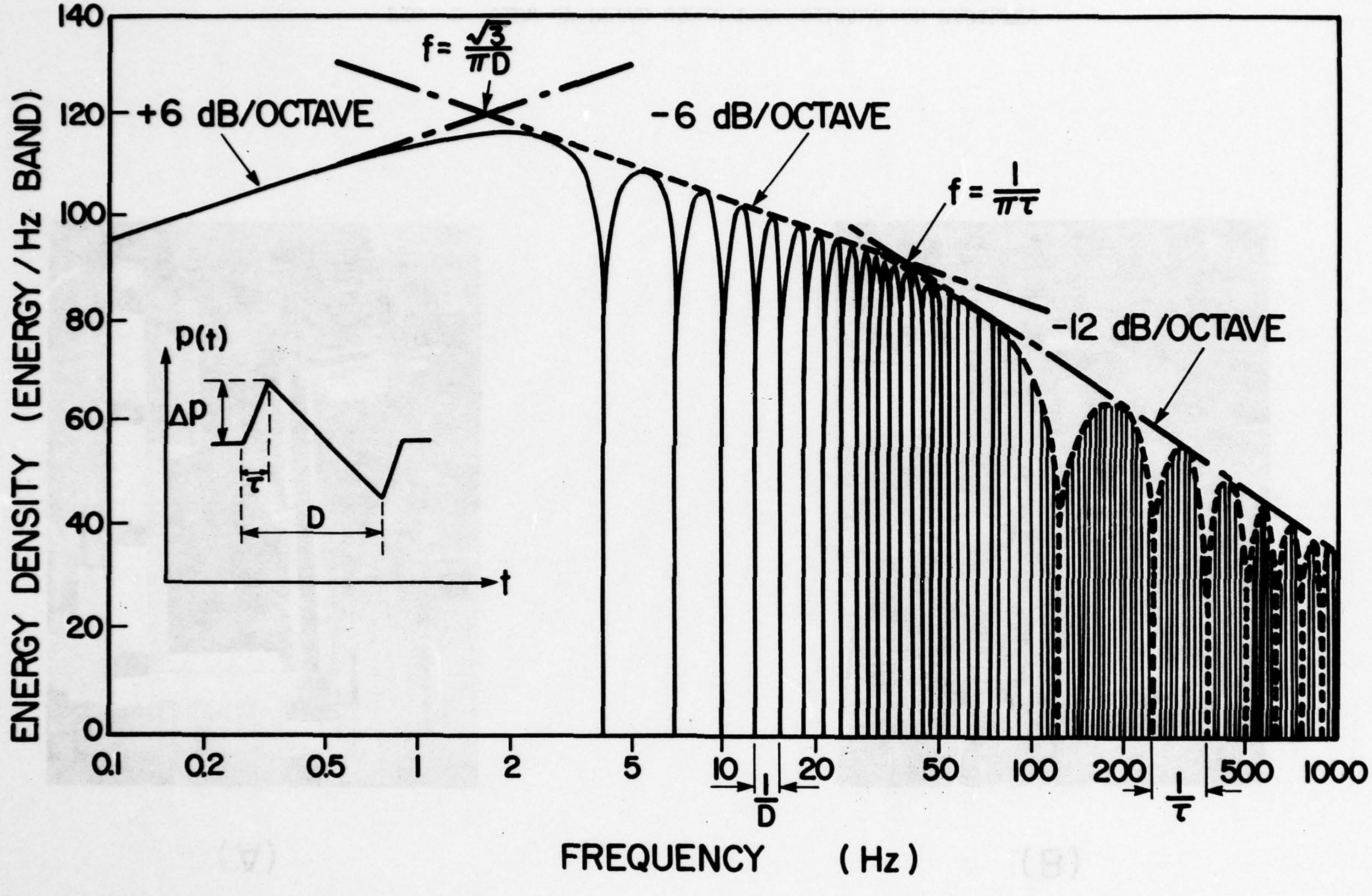
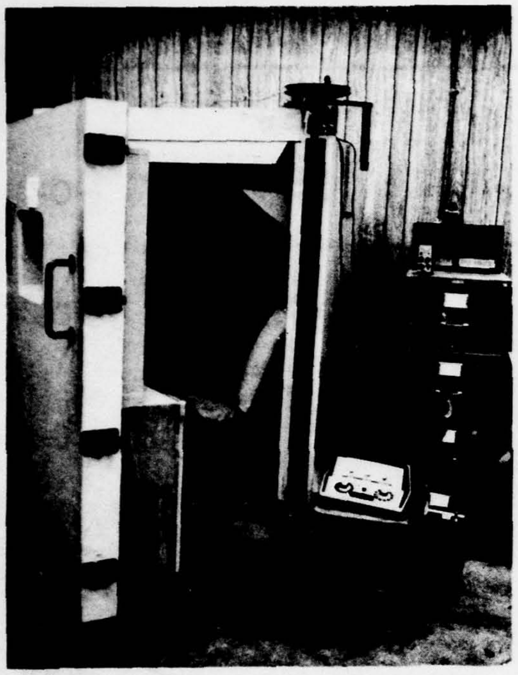


FIG. 1 ENERGY SPECTRAL DENSITY OF SONIC BOOM (FROM REF. 2):
$\Delta p = 50\ N/m^2$, $D = 350$ ms, $\tau = 8$ ms.

(A)



(B)

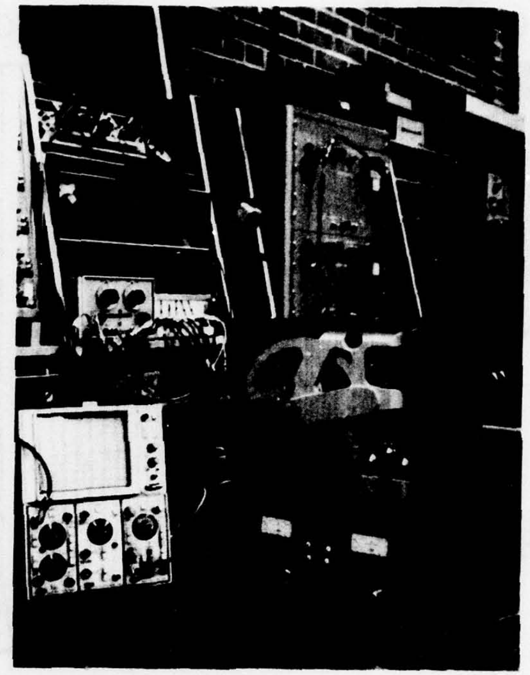


FIG. 2 VIEW OF UTIAS SONIC BOOM SIMULATION FACILITY
(A) Air-tight chamber, (B) Electronic network and equipment.

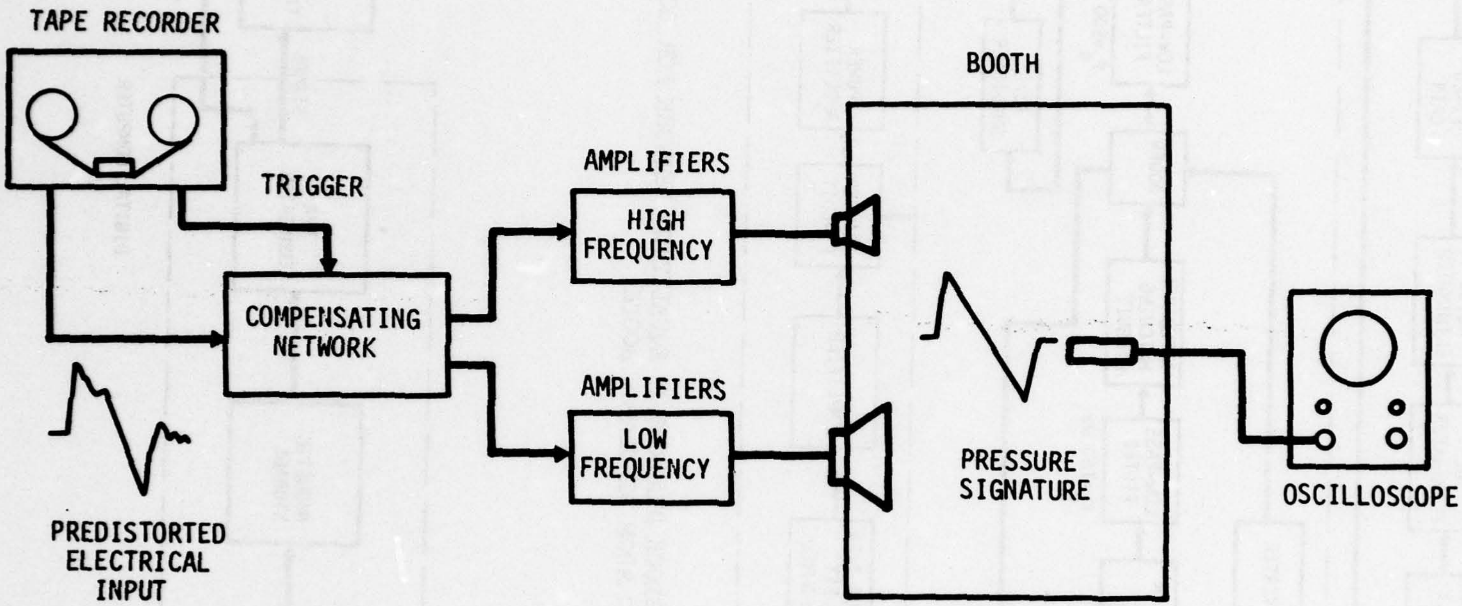


FIG. 3 SCHEMATIC ARRANGEMENT OF UTIAS LOUDSPEAKER-DRIVEN SONIC BOOM SIMULATION BOOTH. COMPENSATORY "PREDISTORTION" OF ELECTRICAL INPUT SIGNAL (SEE TEXT) WAS USED ONLY IN TWO PARTS OF THE PROJECT.

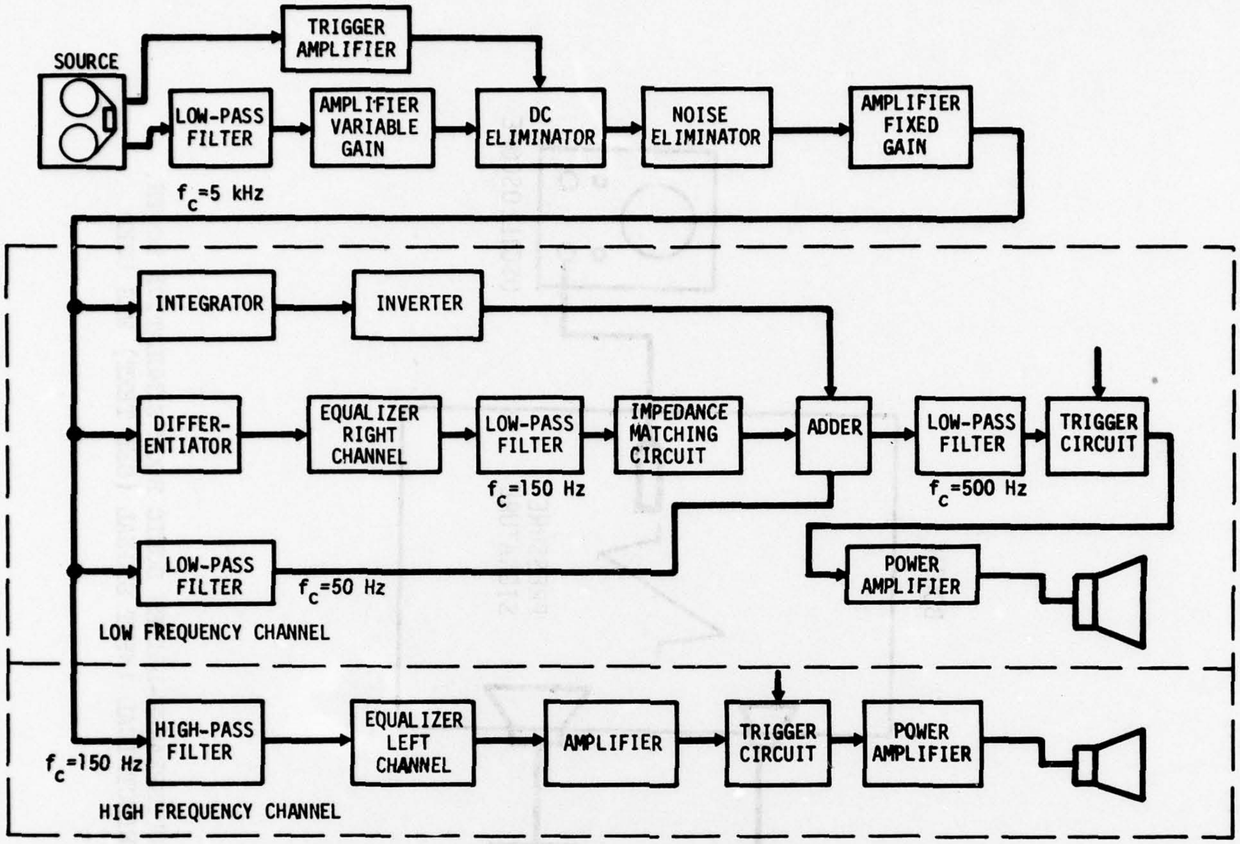


FIG. 4 SIMPLIFIED BLOCK DIAGRAM OF EQUALIZING NETWORK FOR LOUDSPEAKER-DRIVEN SONIC BOOM SIMULATOR BOOTH.

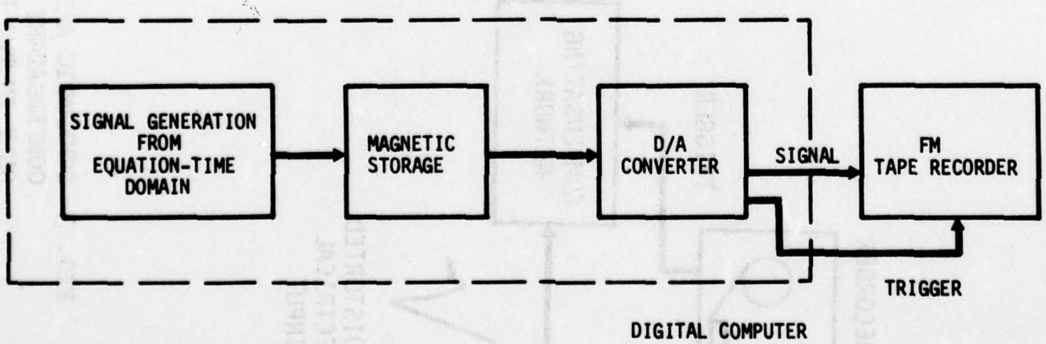


FIG. 5 FLOW CHART FOR TEST SIGNAL GENERATION WITHOUT PREDISTORTION PROCEDURE.

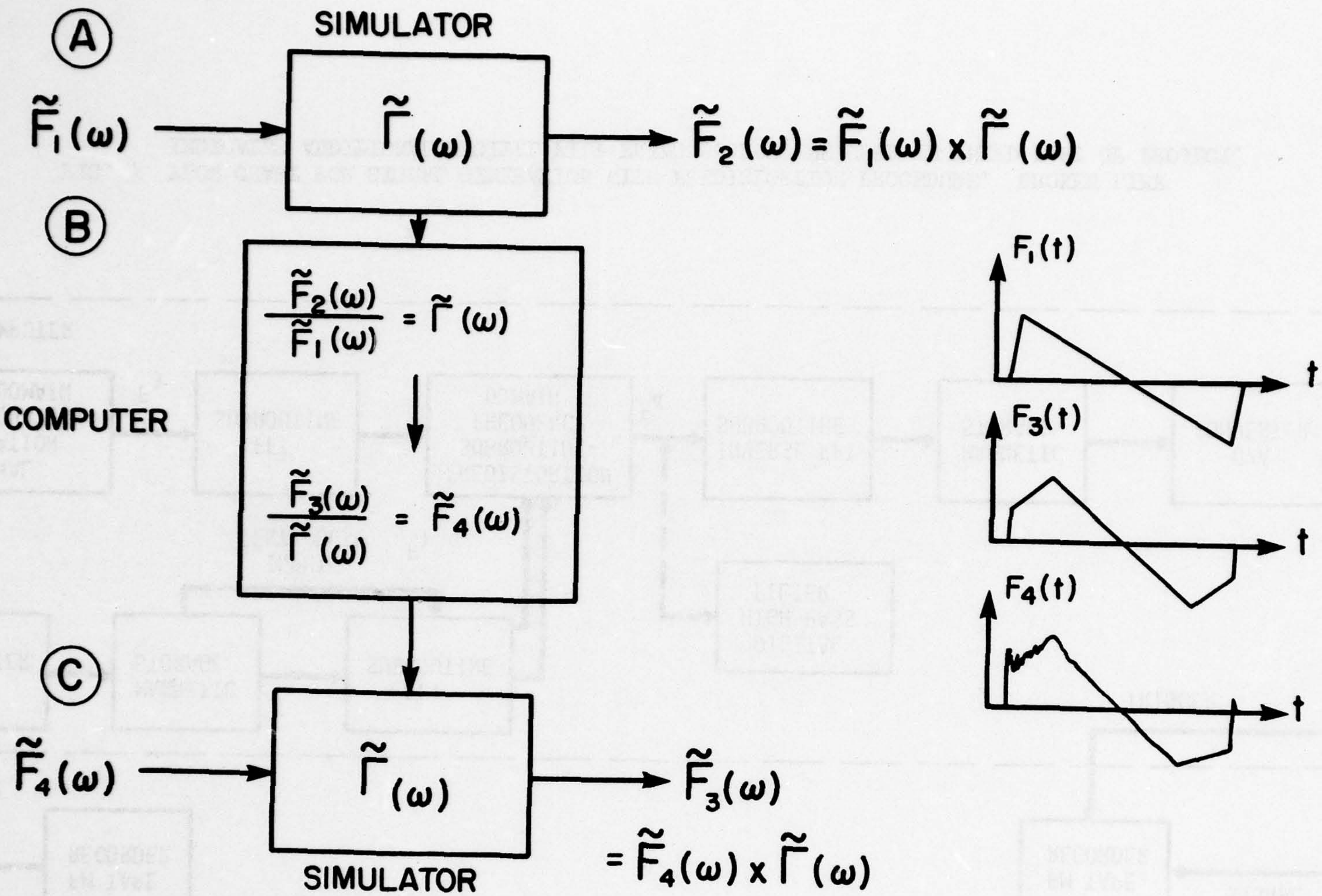


FIG. 6 PREDISTORTION PROCEDURE:

(A) Transfer function measurement using signal $F_1(t)$ (see RH inset)
(B) Computation of transfer function of system $\tilde{\Gamma}(\omega)$ and "predistortion" of test signal $F_3(t)$
(C) Reproduction of "predistorted" signal $F_4(t)$ through the system

Computer performs implied direct and inverse FFT operations $F_1(t) \rightarrow \tilde{F}_1(\omega)$, etc, at appropriate points.

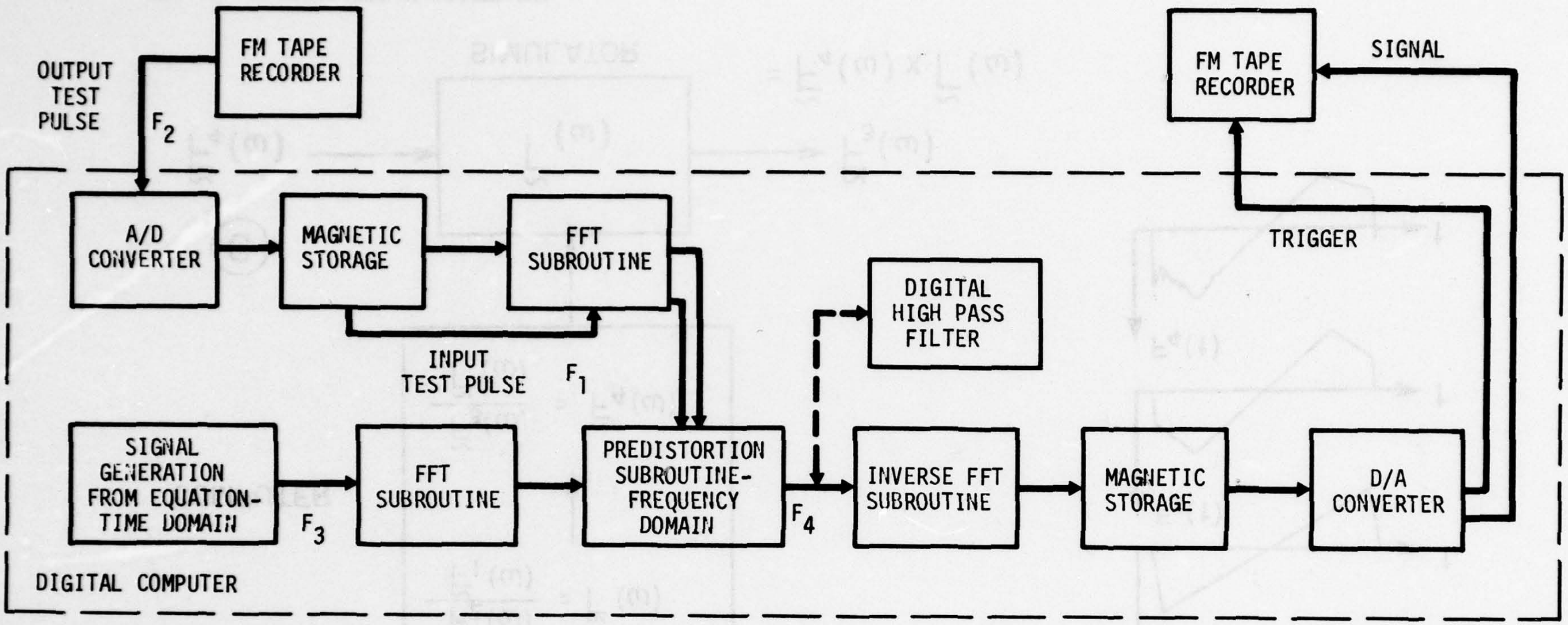


FIG. 7 FLOW CHART FOR SIGNAL GENERATION WITH PREDISTORTION PROCEDURE. BROKEN LINE INDICATES ADDITIONAL DIGITAL FILT ERING SUBROUTINE USED IN THIRD PART OF PROJECT.

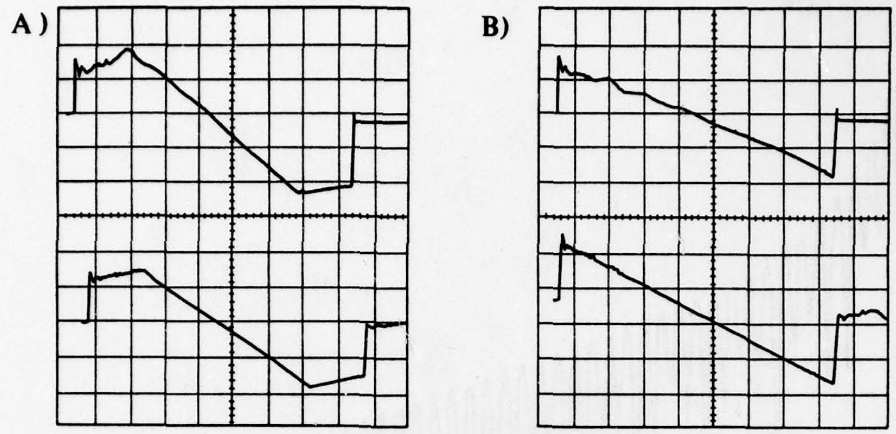


FIG. 8 EFFECT OF "PREDISTORTION" OF ELECTRICAL INPUT SIGNALS (TOP) TO UTIAS SONIC BOOM IN ACHIEVING DESIRED WAVEFORMS (BOTTOM) RECORDED BY MICROPHONE IN BOOTH.

(A) "Low-boom" sonic boom: τ = 1 ms, D = 150 ms, D_1 = 30 ms, x = 0.8
(B) N-wave: τ = 1 ms, D = 150 ms

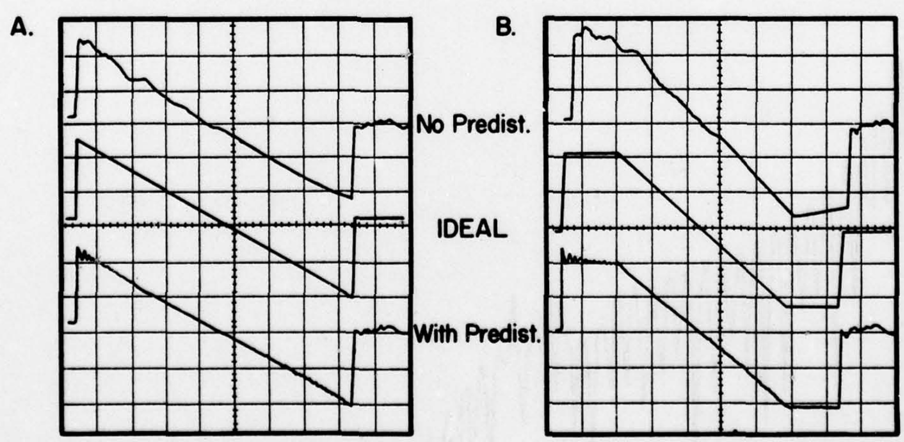


FIG. 9 EXAMPLES OF PRESSURE SIGNALS RECORDED BY MICROPHONE IN UTIAS SIMULATION BOOTH WITHOUT (TOP) AND WITH (BOTTOM) PREDISTORTION OF INPUT SIGNAL.

(A) N-wave: τ = 1 ms; D = 150 ms; Δp = 1 psf
(B) "Low-boom" sonic-boom: τ = 1 ms; D = 150 ms; D_1 = 30 ms; x = 1.0; Δp_{max} = 1 psf.

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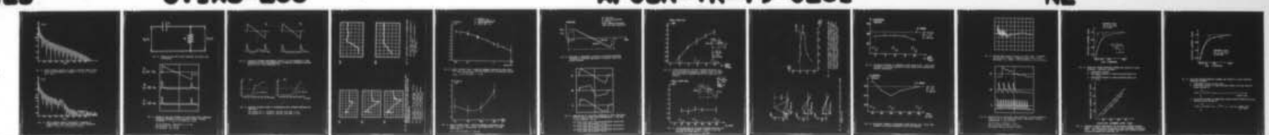

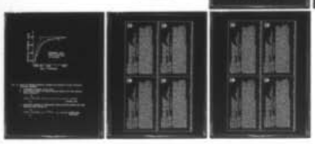

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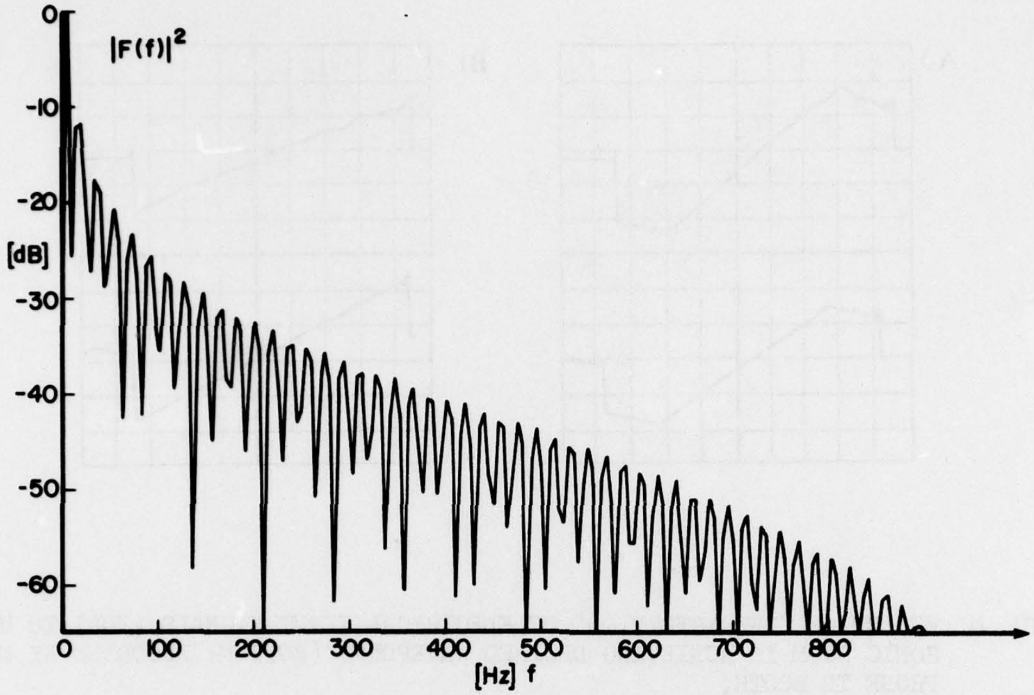

FIG. 10 ENERGY SPECTRAL DENSITY OF "IDEAL" (STRAIGHT LINES) N-WAVE SONIC-BOOM CALCULATED BY COMPUTER (τ = 1 ms, D = 150 ms, Δp = 1 psf).

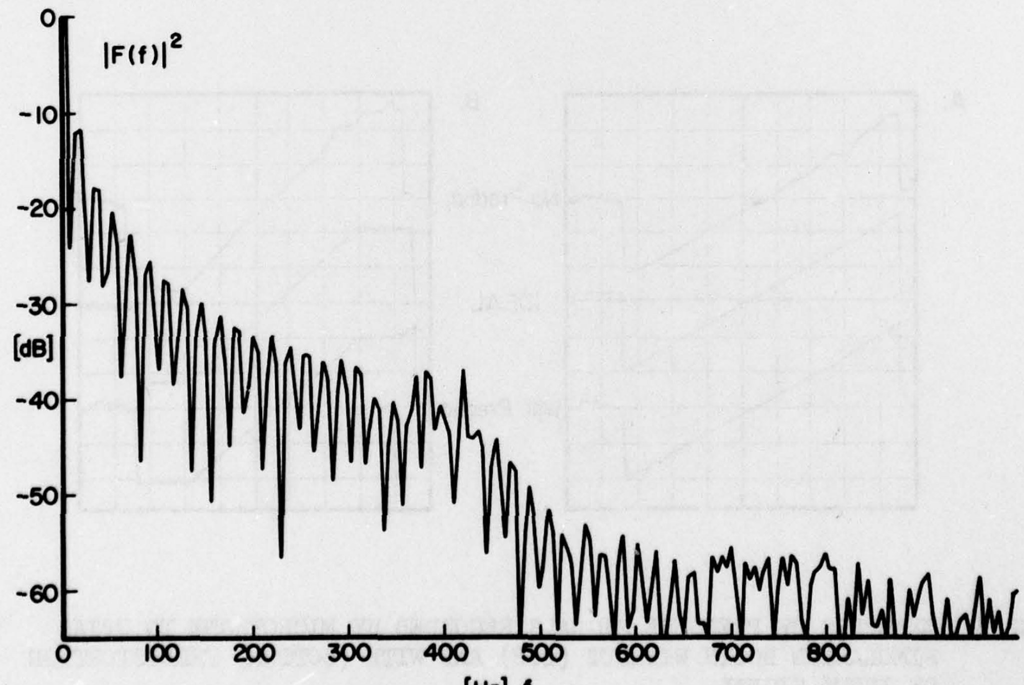

FIG. 11 ENERGY SPECTRAL DENSITY CALCULATED BY COMPUTER OF "PREDISTORTED" SONIC BOOM RECORDED IN UTIAS SIMULATION BOOTH (τ = 1 ms, D = 150 ms, Δp = 1 psf).

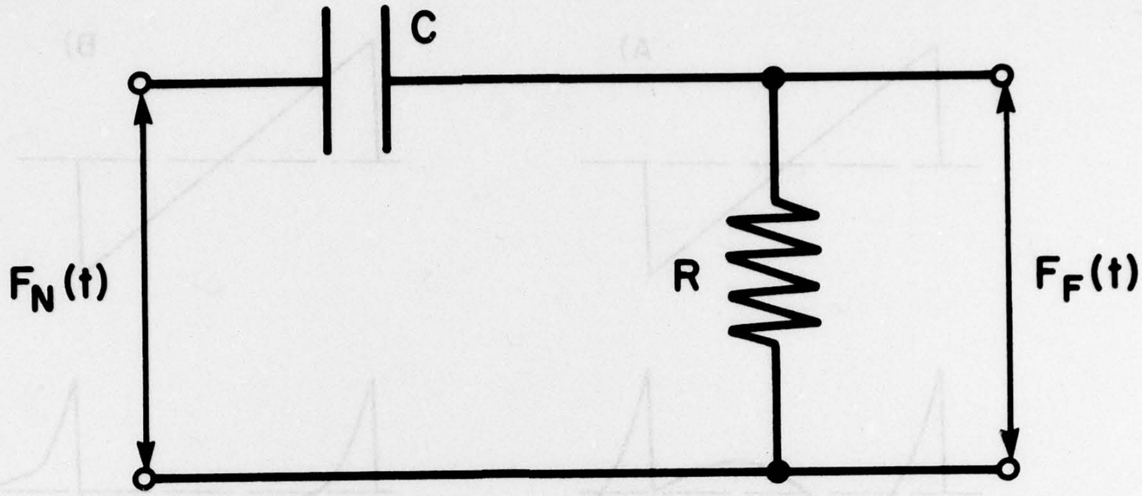

FIG. 12 SIMPLE RC HIGH PASS FILTER SIMULATED IN DIGITAL FORM IN COMPUTER.

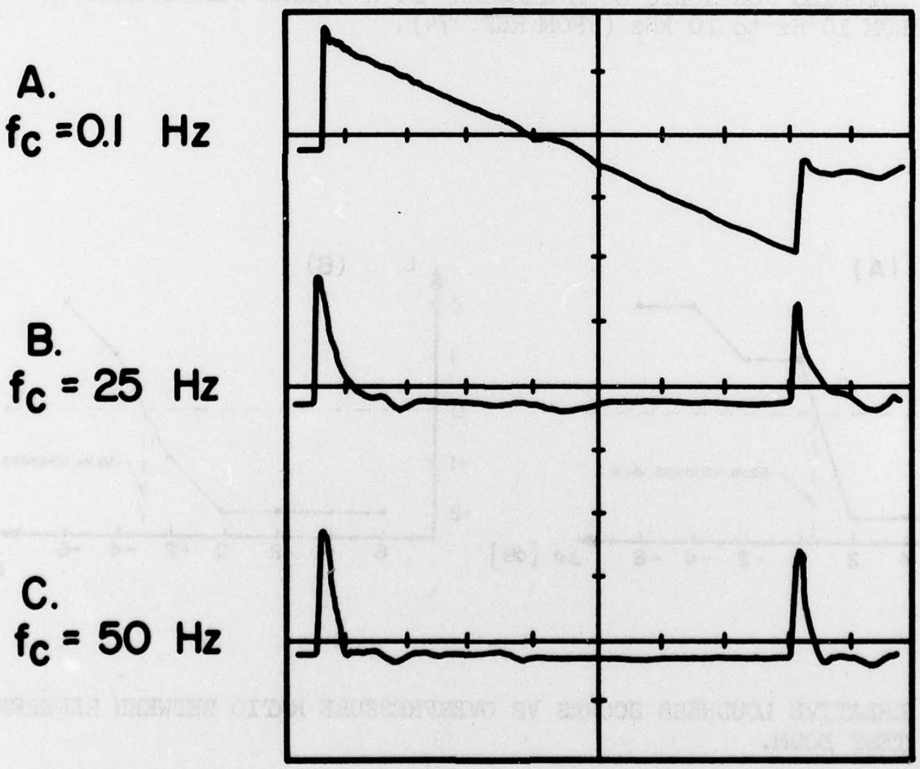

FIG. 13 EFFECT OF HIGH PASS FILTERING ON SONIC BOOM SIGNAL (PRESSURE WAVEFORMS RECORDED IN UTIAS SIMULATION BOOTH). N-WAVE SIGNATURE: τ = 1 ms; D = 150 ms; Δp = 1 psf.

(A) No filtering - effective f_c = 0.1 Hz
(B) Filtered - f_c = 25 Hz
(C) Filtered - f_c = 50 Hz

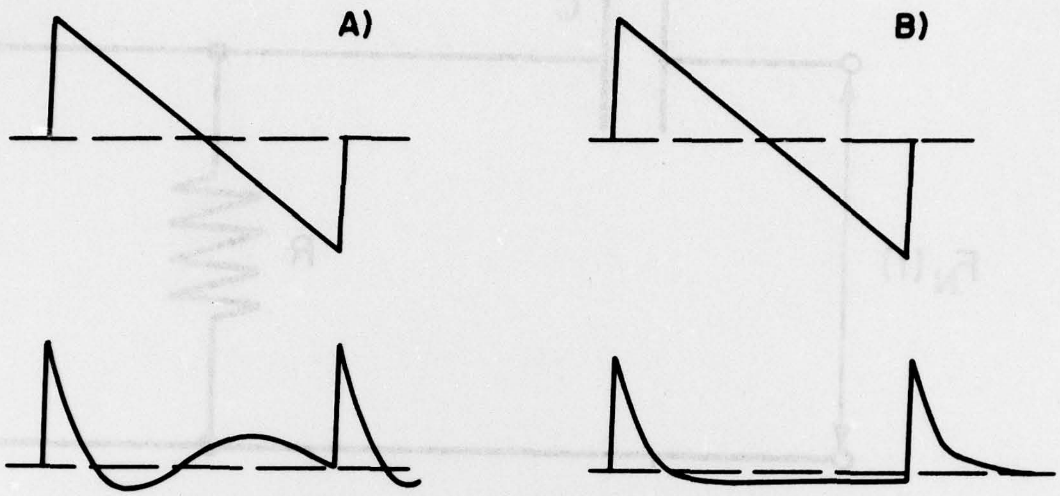

FIG. 14 COMPARISON BETWEEN EXPERIMENTAL (ANALOG) (A) AND PREDICTED (B) TIME HISTORIES FOR SONIC BOOM MEASURED BY A SYSTEM WITH FREQUENCY RESPONSE FROM 10 Hz to 10 kHz (FROM REF. 74).

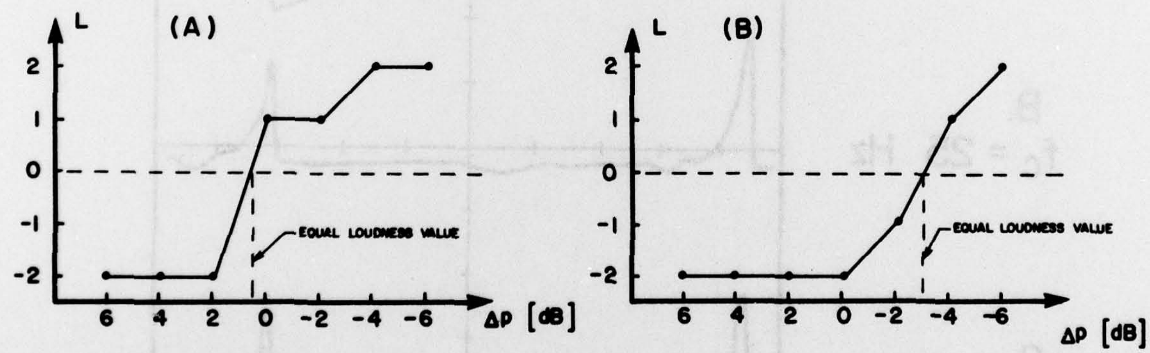

FIG. 15 RELATIVE LOUDNESS SCORES VS OVERPRESSURE RATIO BETWEEN REFERENCE AND TEST BOOM.

(A) Subject No. 4 - duration = 200 ms; rise time = 1 ms
(B) Subject No. 4 - duration = 200 ms; rise time = 0.5 ms

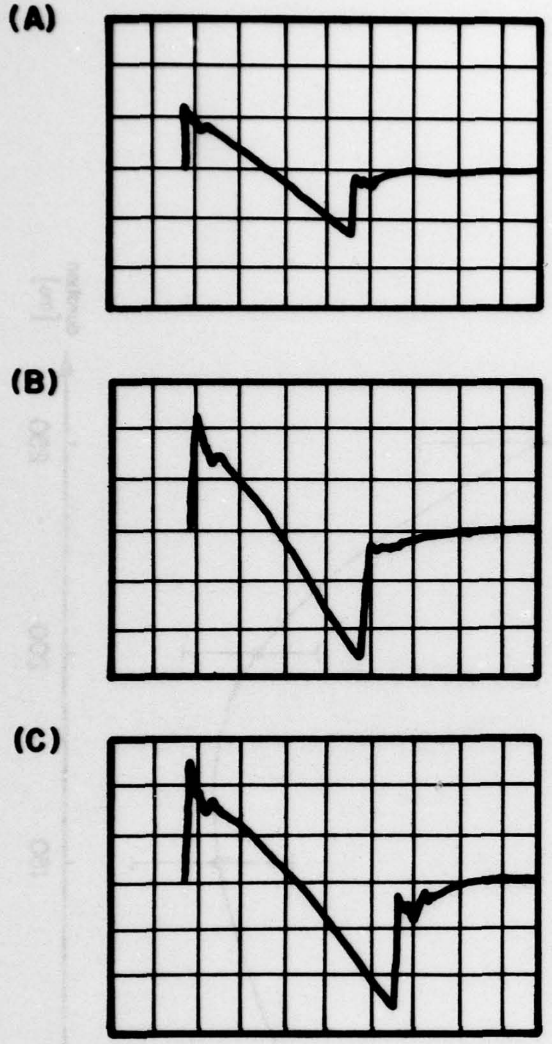

FIG. 16 REPRODUCTION OF N-WAVE SONIC-BOOM SIGNATURES BY UTIAS SIMULATION BOOTH AS USED IN FIRST SERIES OF TESTS (NO PREDISTORTION).

(A) Rise time = 0.22 ms; duration = 200 ms; overpressure = 1 psf (48 N/m^2)
(B) Rise time = 10 ms; duration = 200 ms; overpressure = 2 psf (96 N/m^2)
(C) Rise time = 1 ms; duration = 250 ms; overpressure = 2 psf (96 N/m^2)

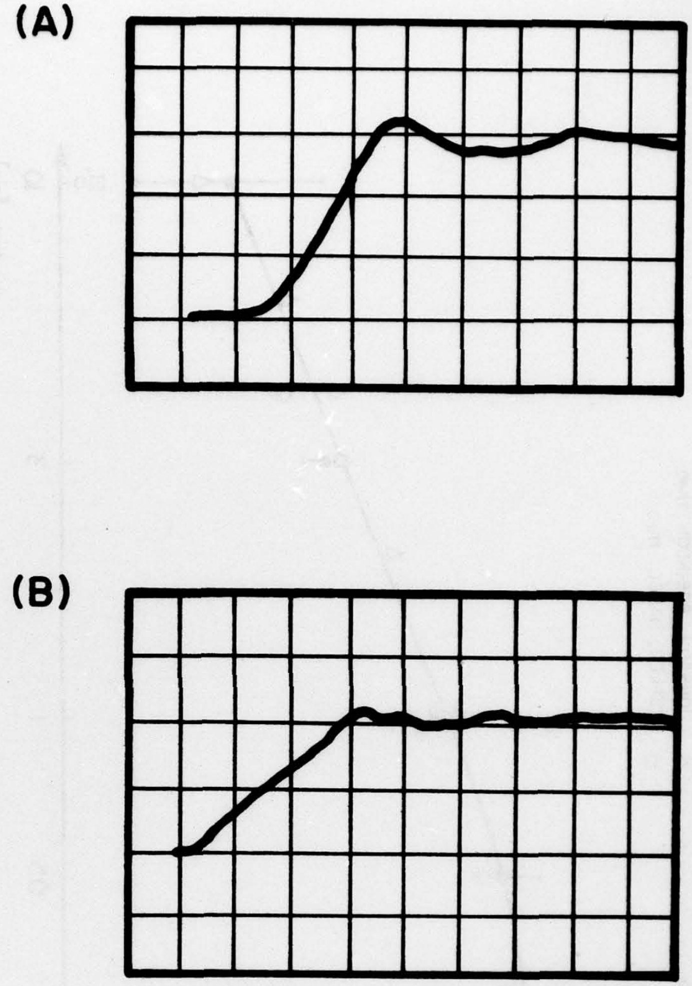

FIG. 17 FRONT SHOCK OF FIG. 16 TO EXPANDED SCALE.

(A) Rise time = 0.22 ms; overpressure = 1 psf (48 N/m^2)
(B) Rise time = 3 ms; overpressure = 1 psf (48 N/m^2)

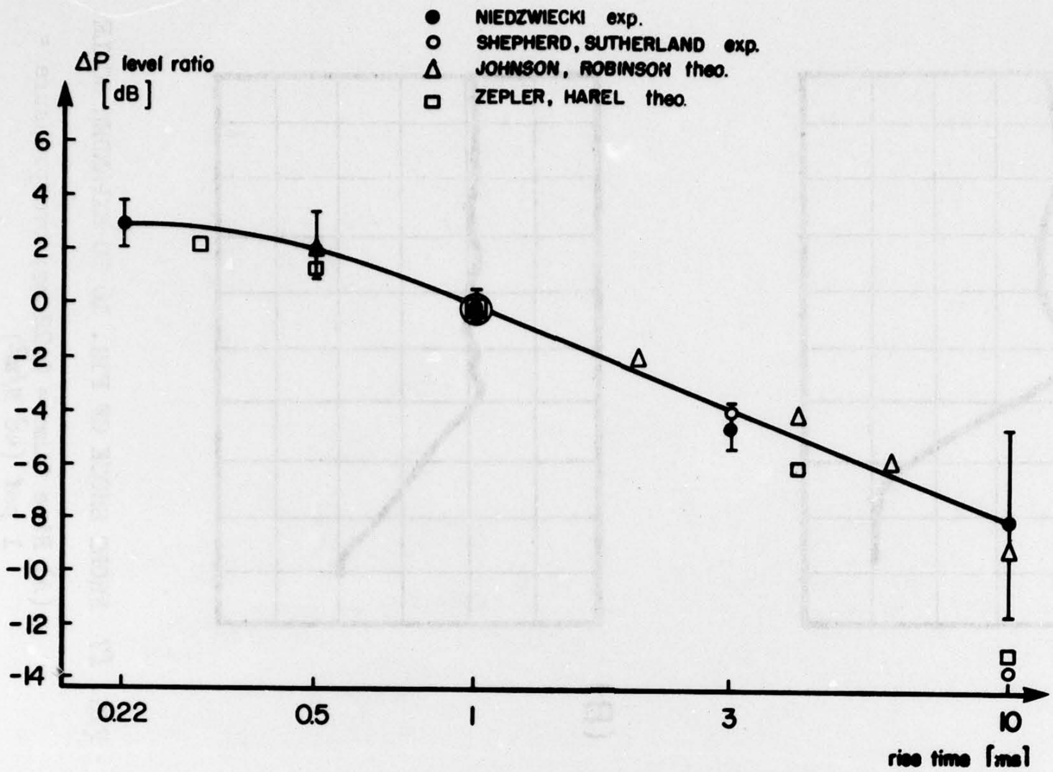

FIG. 18 EQUAL LOUDNESS CURVE. TRADE-OFF BETWEEN OVERPRESSURE LEVEL RATIO ($-20 \log_{10} \Delta p_{test}/\Delta p_{ref}$) AND RISE TIME FOR 200 ms DURATION N-WAVES.
Δp_{ref} = 1 psf (48 N/m^2), τ = 1 ms.

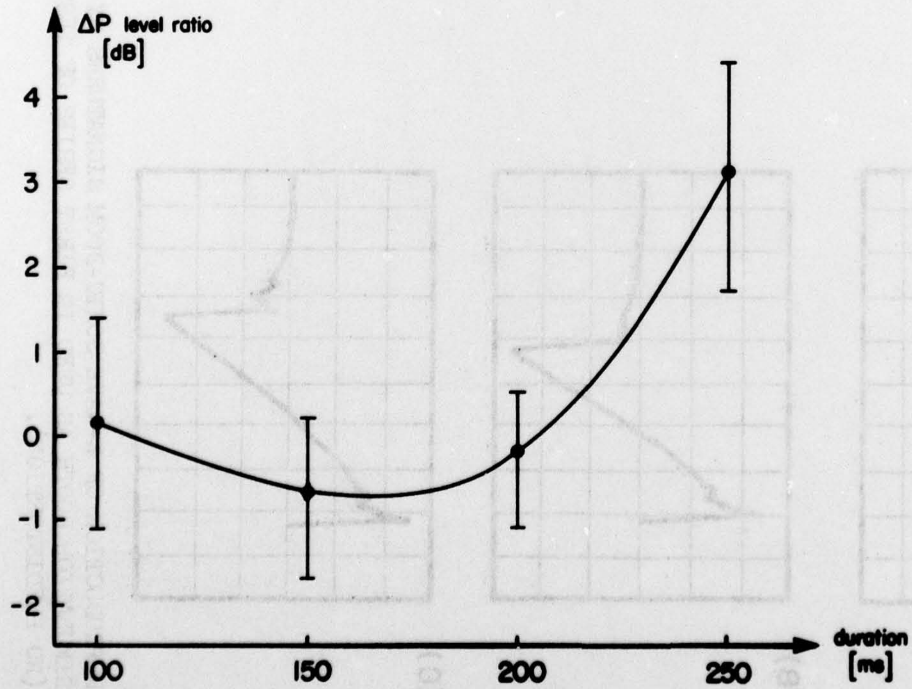

FIG. 19 EQUAL LOUDNESS CURVE. TRADE-OFF BETWEEN OVERPRESSURE LEVEL RATIO ($-20 \log_{10} \Delta p_{test}/\Delta p_{ref}$) AND DURATION FOR 1 ms RISE TIME N-WAVE.
Δp = 1 psf (48 N/m^2), D = 200 ms.

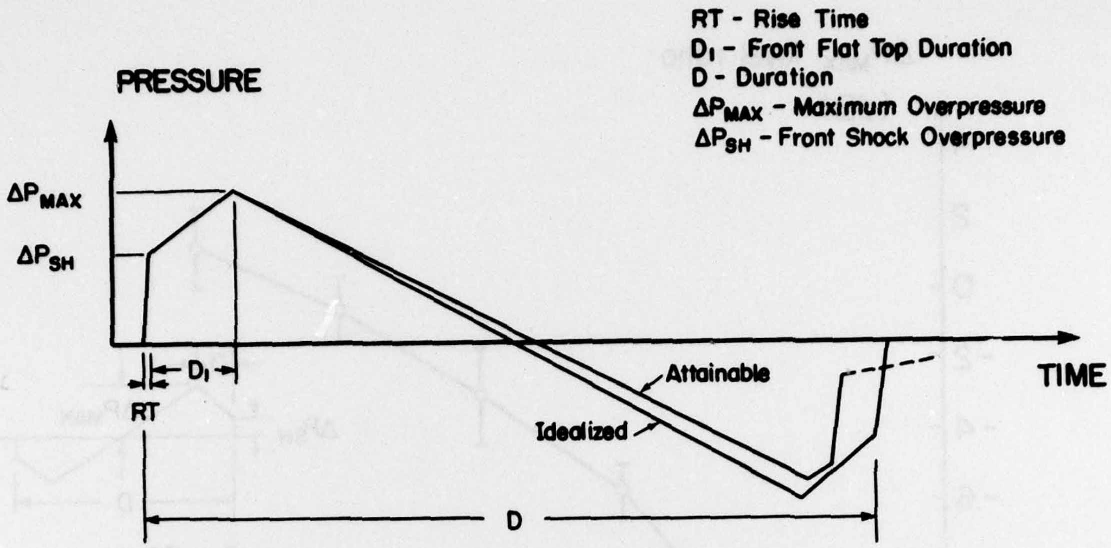

FIG. 20 IDEALIZED VS ATTAINABLE "LOW-BOOM" SONIC-BOOM SIGNATURES. "ATTAINABLE" SIGNIFIES REALIZABLE VIA AIRCRAFT DESIGN AND FLIGHT PROCEDURE.

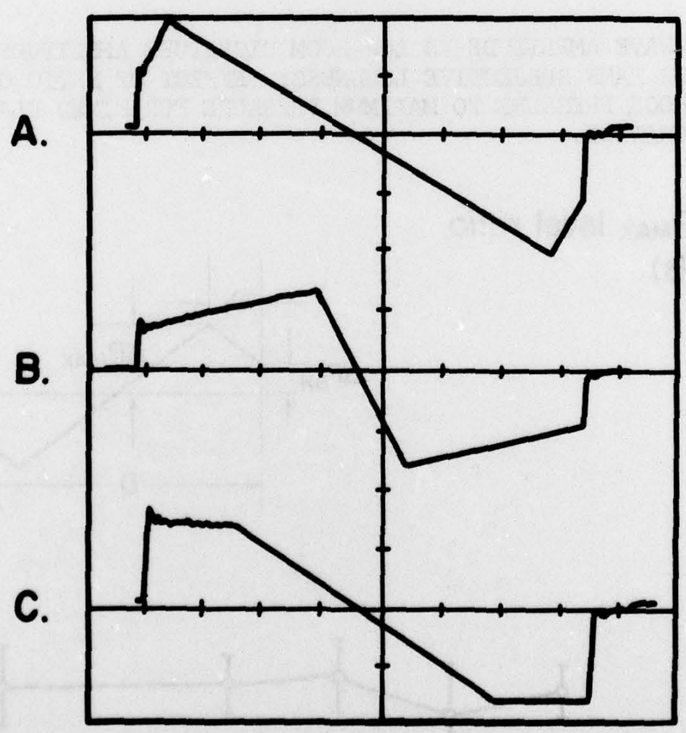

FIG. 21 REPRODUCTION OF "LOW-BOOM" SIGNATURES BY UTIAS SIMULATION BOOTH AS USED IN TESTS (WITH PREDISTORTION). RISE TIME = 1 ms, DURATION = 150 ms.

(A) Ratio front shock overpressure/maximum overpressure = 0.5; flat top duration = 10 ms

(B) Ratio front shock overpressure/maximum overpressure = 0.5; flat top duration = 60 ms

(C) Ratio front shock overpressure/maximum overpressure = 1.0; flat top duration = 30 ms

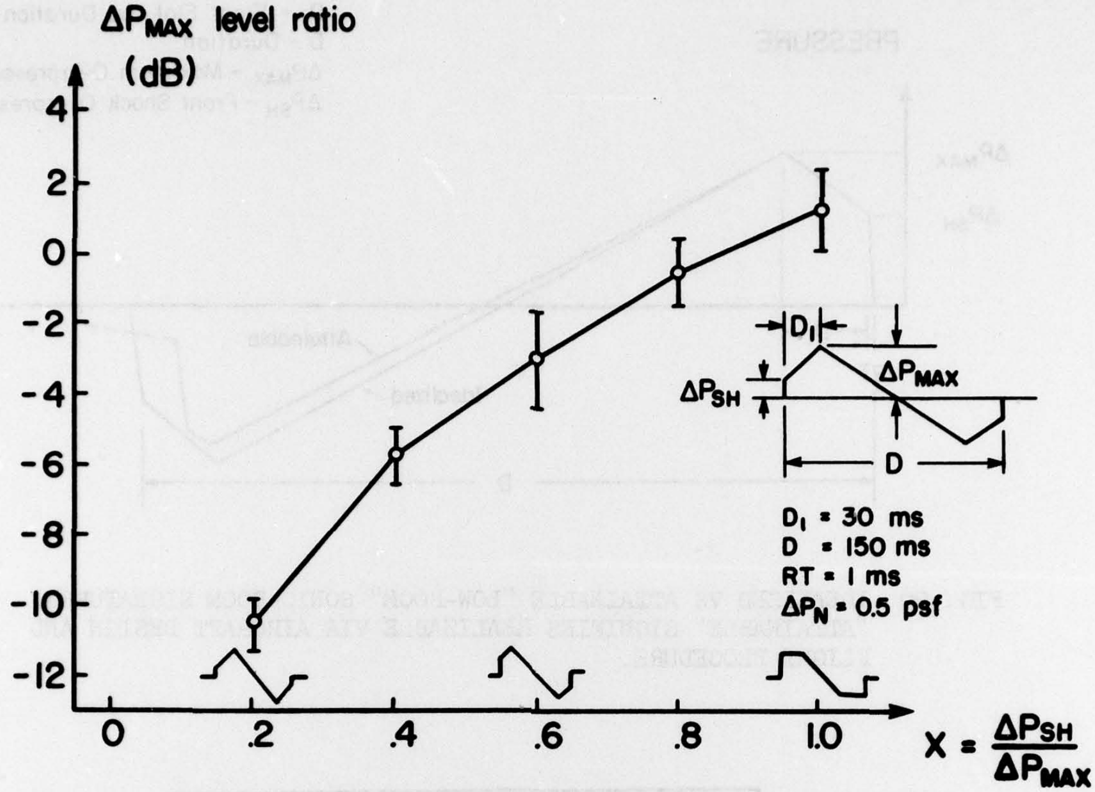

FIG. 22 N-WAVE AMPLITUDE VS LOW-BOOM SIGNATURE AMPLITUDE (dB) FOR SAME SUBJECTIVE LOUDNESS: EFFECT OF RATIO OF FRONT SHOCK PRESSURE TO MAXIMUM PRESSURE FOR FIXED FLAT TOP DURATION.

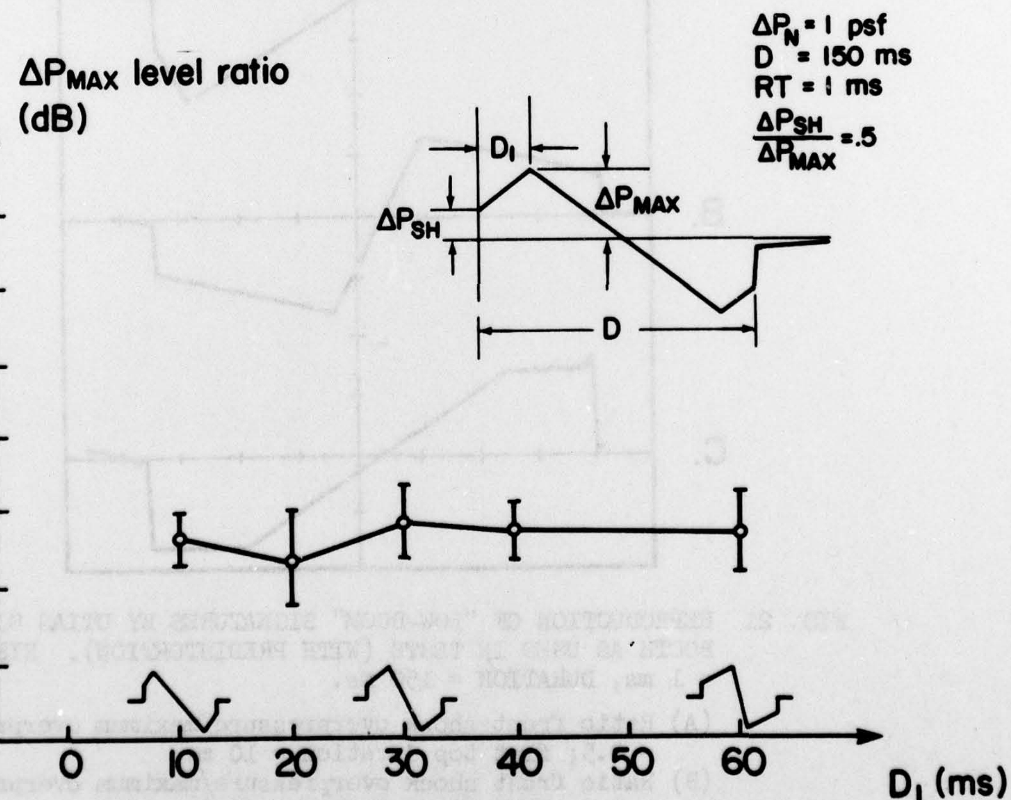

FIG. 23 N-WAVE AMPLITUDE VS LOW-BOOM SIGNATURE AMPLITUDE (dB) FOR SAME SUBJECTIVE LOUDNESS: EFFECT OF FLAT TOP DURATION FOR FIXED RATIO $x = p_{SH}/p_{MAX}$.

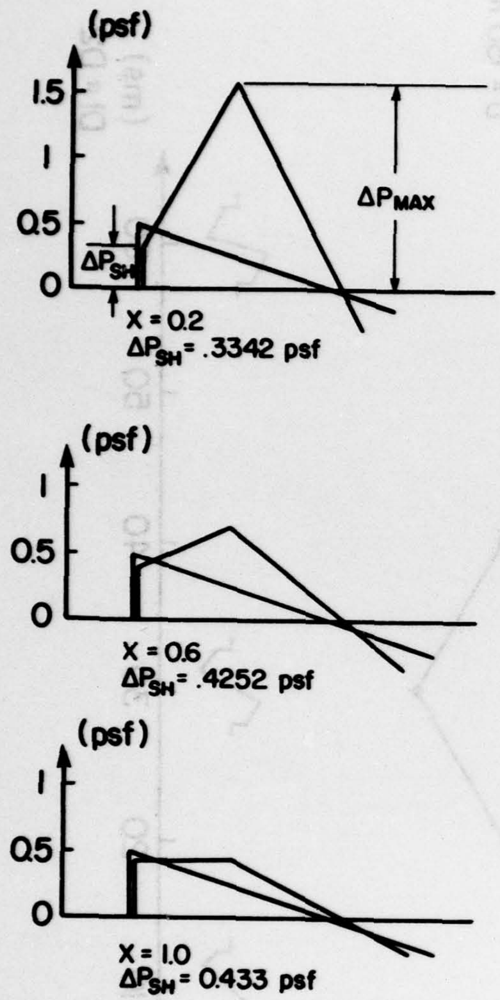

FIG. 24 SHAPES OF SOME PAIRS (LOW-BOOM AND N-WAVE) JUDGED EQUALLY LOUD.

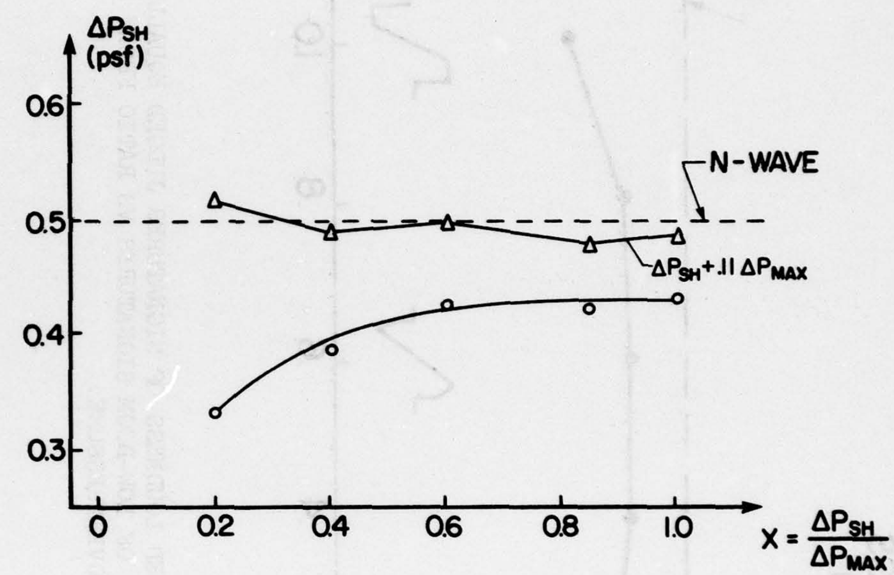

FIG. 25 LOW-BOOM SIGNATURE FRONT SHOCK OVERPRESSURE FOR SAME SUBJECTIVE LOUDNESS AS N-WAVE WITH OVERPRESSURE p_N = 0.5 psf (24 N/m^2): (CIRCLES) EFFECT OF RATIO FRONT SHOCK OVERPRESSURE/MAXIMUM OVERPRESSURE FOR FIXED FLAT TOP DURATION; (TRIANGLES) CALCULATED VALUES OF "EFFECTIVE OVERPRESSURE" OF EQUIVALENT N-WAVE.

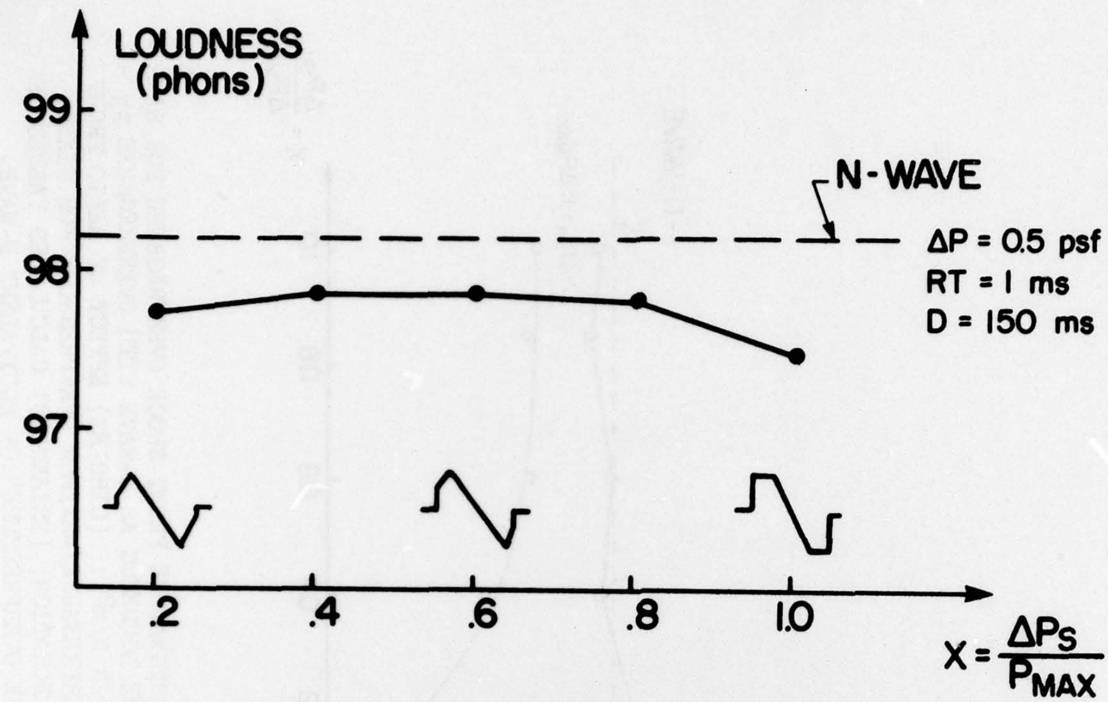

FIG. 26 CALCULATED LOUDNESS OF SIGNATURES JUDGED EQUALLY LOUD. SOLID LINE: LOUDNESS OF LOW-BOOM SIGNATURES VS RATIO FRONT SHOCK OVERPRESSURE/ MAXIMUM OVERPRESSURE.

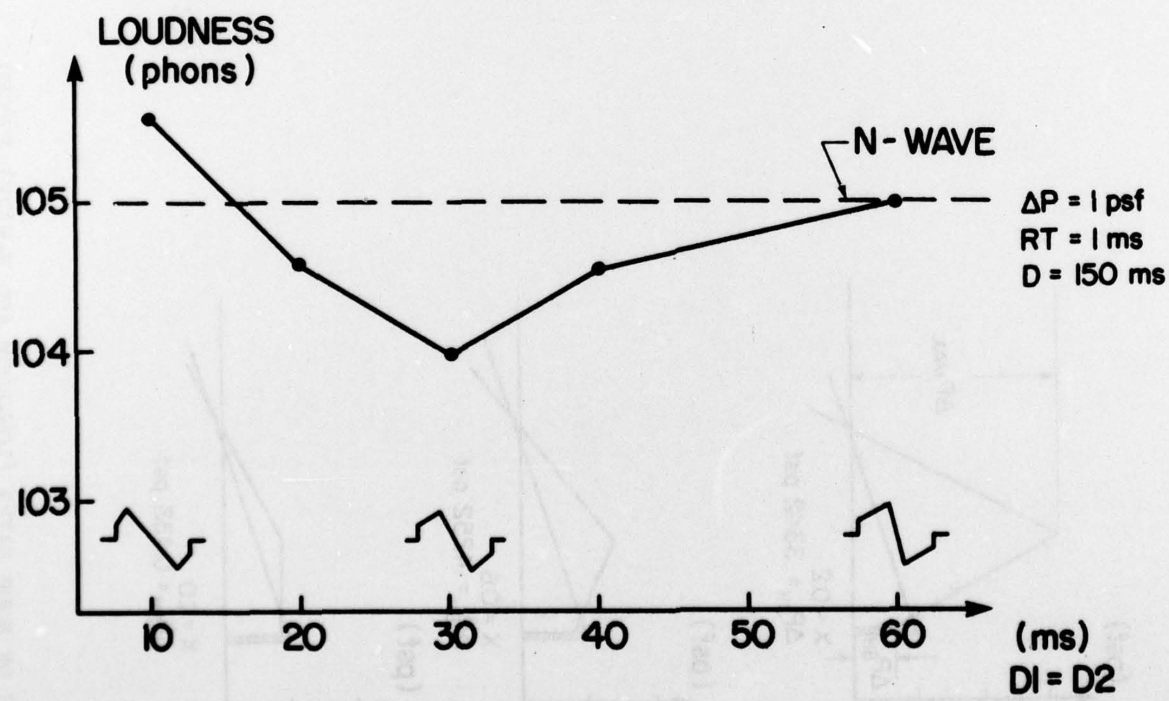

FIG. 27 CALCULATED LOUDNESS OF SIGNATURES JUDGED EQUALLY LOUD. SOLID LINE: LOUDNESS OF LOW-BOOM SIGNATURES VS FLAT TOP DURATION.

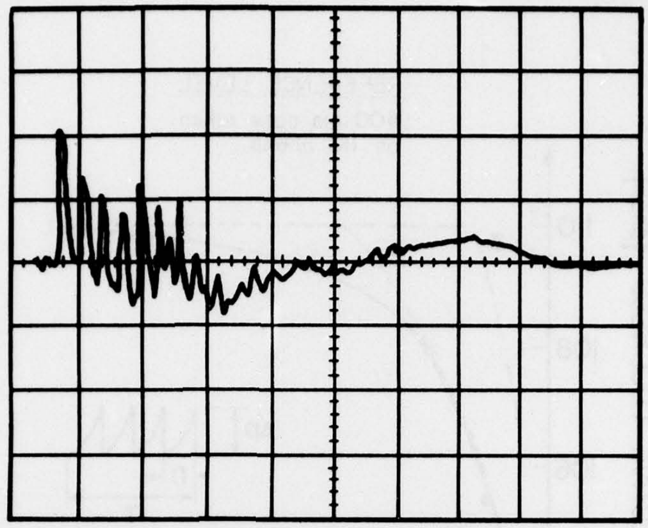

FIG. 28 EXAMPLE TIME-PRESSURE HISTORY OF QUARRY BLAST. SIGNATURE WAS RECORDED AT MILTON LIMESTONE QUARRY, MAY 16, 1974 (FROM REF. 11). SCALE: 1.78 psf/division, 125 ms/division.

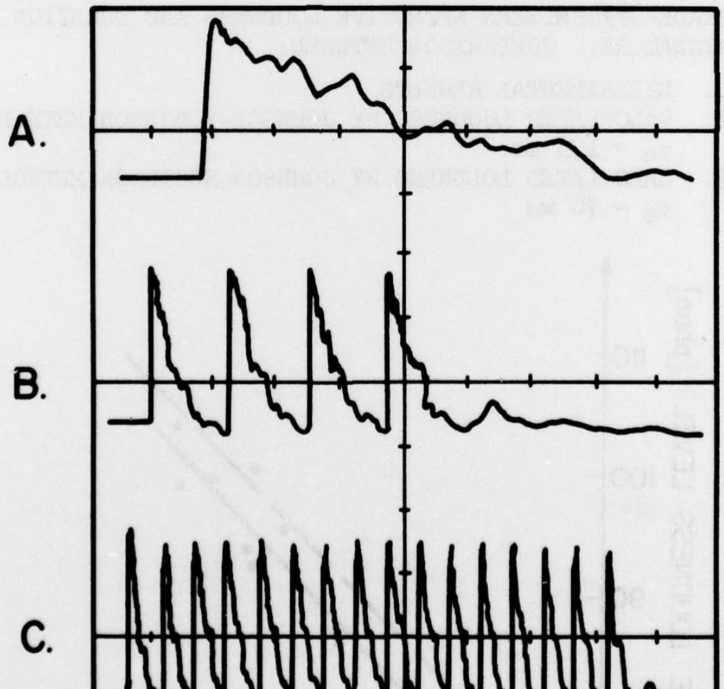

FIG. 29 REPRODUCTION OF "IDEALIZED" BLAST WAVEFORMS BY UTIAS SIMULATION BOOTH AS USED IN THE SUBJECTIVE TEST. RISE TIME = 0.22 ms, OVERPRESSURE = 1 psf (48 N/m^2).

(A) One pulse, duration = 25 ms
(B) Four pulses, duration = 100 ms
(C) Sixteen pulses, duration = 400 ms

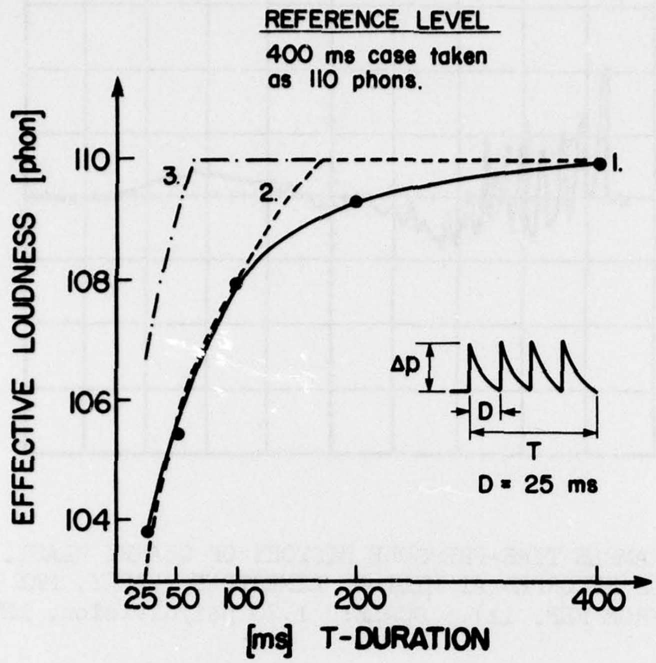

FIG. 30 TRADE-OFF BETWEEN EFFECTIVE LOUDNESS AND DURATION OF BLAST SIGNATURE. COMPARISON BETWEEN:

1. EXPERIMENTAL RESULTS
2. *CALCULATED LOUDNESS* BY *JOHNSON-ROBINSON METHOD FOR* τ_c = 175 ms
3. CALCULATED LOUDNESS BY JOHNSON-ROBINSON METHOD FOR τ_c = 70 ms

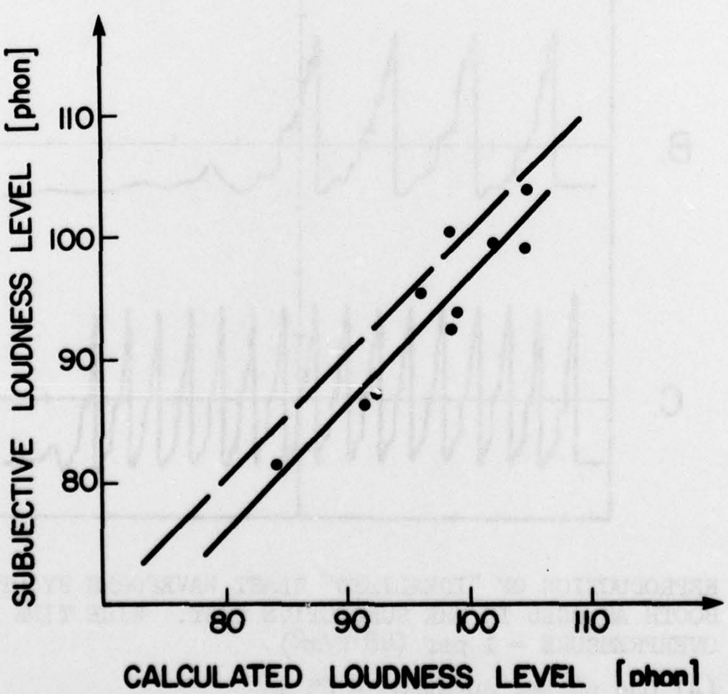

FIG. 31 COMPARISON BETWEEN EXPERIMENTAL AND CALCULATED (JOHNSON-ROBINSON METHOD) LOUDNESS LEVELS FOR THE HALF N-WAVE SIGNATURES (CONTINUOUS LINE). IDEAL CORRELATION BETWEEN CALCULATED AND SUBJECTIVE RESULTS IS SHOWN BY BROKEN LINE (FROM REF. 63).

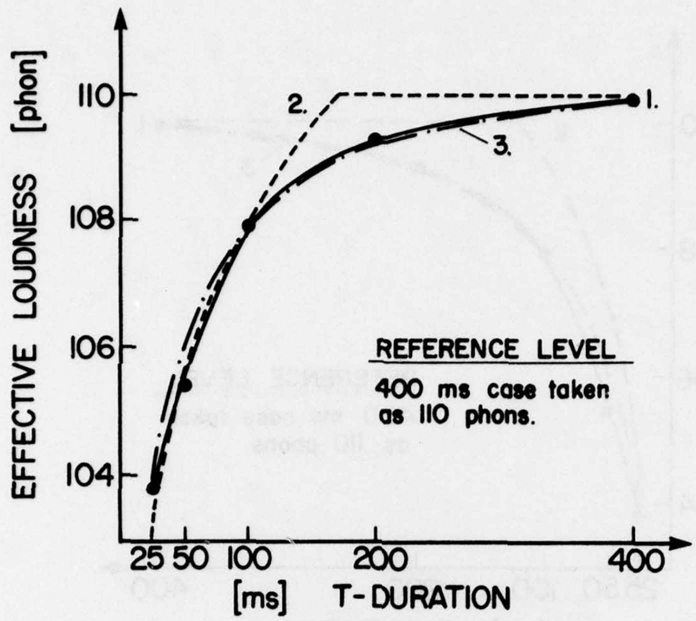

FIG. 32 TRADE-OFF BETWEEN EFFECTIVE LOUDNESS AND DURATION OF BLAST SIGNATURE. COMPARISON BETWEEN:

1. EXPERIMENTAL RESULTS (SOLID LINE)
2. CALCULATED LOUDNESS BY JOHNSON-ROBINSON METHOD WITH BAND PRESSURE LEVELS DEFINED AS

$$(1/\pi T) \int_{\omega_1}^{\omega_2} |F(\omega)|^2 d\omega \times (T/\tau_c,\ T \leq \tau_c;\ 1,\ T \geq \tau_c) \quad (\tau_c = 175 \text{ ms})$$

(BROKEN LINE)

3. CALCULATED LOUDNESS BY GENERALIZED JOHNSON-ROBINSON METHOD WITH BAND PRESSURE LEVELS DEFINED AS

$$(1/\pi T) \int_{\omega_1}^{\omega_2} |F(\omega)|^2 d\omega \times [1 - e^{-T/\tau_c}] \quad (\tau_c = 125 \text{ ms})$$

(BROKEN LINE WITH DOTS)

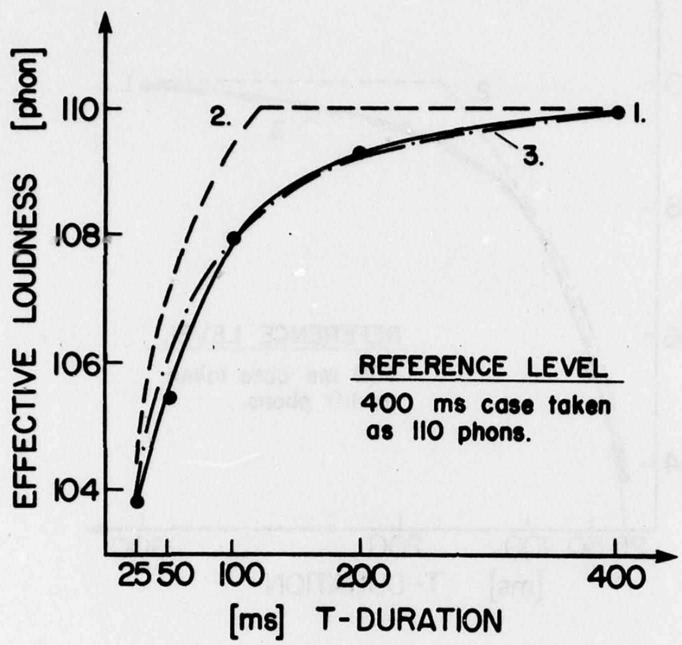

FIG. 33 TRADE-OFF BETWEEN EFFECTIVE LOUDNESS AND DURATION OF BLAST SIGNATURE. COMPARISON BETWEEN:

1. EXPERIMENTAL RESULTS (SOLID LINE)
2. CALCULATED LOUDNESS BY JOHNSON-ROBINSON METHOD WITH BAND PRESSURE LEVELS DEFINED AS

$$(1/\pi T) \int_{\omega_1}^{\omega_2} |F(\omega)|^2 d\omega \;\; x\,(T/\tau_c,\ T \leq \tau_c;\ 1,\ T \geq \tau_c) \;\; (\tau_c = 125 \text{ ms})$$

(BROKEN LINE)

3. CALCULATED LOUDNESS BY GENERALIZED JOHNSON-ROBINSON METHOD WITH BAND PRESSURE LEVELS DEFINED AS

$$(1/\pi T) \int_{\omega_1}^{\omega_2} |F(\omega)|^2 d\omega \;\; x\,[1 - e^{-T/\tau_c}] \quad (\tau_c = 125 \text{ ms})$$

(BROKEN LINE WITH DOTS)

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